THE BELL SYSTEM
TECHNICAL JOURNAL

VOLUME XXXIX May 1960 NUMBER 3

Capabilities of the Telephone Network for Data Transmission

By A. A. ALEXANDER, R. M. GRYB and D. W. NAST

(Manuscript received January 22, 1960)

This paper presents the results of a nationwide data transmission field testing program on the telephone switched message network. Error performance using the FM digital subset is described and basic transmission characteristics such as net loss, bandwidth, envelope delay and noise are given.

I. INTRODUCTION

The telephone industry has a long history of providing data transmission services. Over the years many varieties of service offerings making use of the range from narrow-band telegraph[1] channels up to 4-mc video channels have been provided. Such services usually have been provided on a private line basis by adapting regular telephone facilities to the particular data service requirement.

The increased use of computers and automatic data processing systems in the commercial, industrial and military areas has substantially increased the demand for greater varieties of data services and data transmission channels. This expansion, with its attendant requirement for a variety of speeds and channel usage time, has encouraged development of service offerings that use the regular switched message telephone network in establishing the communication channels. In the Bell System this service concept has been given the name of Data-Phone,* which

* Data-Phone is a trademark of the American Telephone and Telegraph Company identifying Bell System equipment used in this Bell System service.

envisions not one, but a whole family of data transmission systems operating on the regular switched network. It will encompass a broad range of speed capabilities and meet a variety of performance requirements.

Operationally, the Data-Phone service is quite simple. A regular telephone call is made to establish a connection between two points. Usually, regular voice communication may be carried on if required. Operation of a pushbutton associated with the telephone set at each end of the connection disconnects the telephone instruments and connects data subsets to the telephone lines. The subset, depending upon the type, accepts analog or digital (usually binary) information at the transmitting end and, if necessary, modulates the baseband signal to a frequency band suitable for use over telephone circuits. At the receiving end the data subset demodulates the line signal and returns it to baseband. At the end of the transmission regular voice communication can be resumed, if required, or the connection can be terminated by hanging up the telephone set.

Additional operational features can, of course, be built into the Data-Phone service as may be required. For example, machines may be used to dial up the connection, answer back, intercommunicate and disconnect entirely independent of human assistance. These and other similar features are obvious extensions of the Data-Phone concept.

The switched telephone network is designed primarily to handle voice communication. Many of the design criteria are based upon talker and listener habits and preferences. The resulting characteristics, while suitable for data transmission are not as optimized as they are when a communication channel is designed specifically for data use. Frequently it is possible to take advantage of certain speech or human ear characteristics to provide better or more economical service. The uses of compandored[2] carrier systems and echo suppressors[3] are typical examples. When the telephone network is used to provide communication channels for systems having nonhuman characteristics the advantages of these special devices are lost.

In order to use the telephone network for the variety of uses contemplated under the Data-Phone concept, it is necessary to know the following:

i. What is contained in the telephone network and how it operates.

ii. What are the voice and data transmission characteristics of connections in the message network and to what extent they limit the transmission of data signals.

Once these are determined an objective evaluation of the switched message network can be made, and data systems can be designed with a reasonable degree of assurance for successful application.

II. SWITCHED TELEPHONE NETWORK

A great deal of information describing the component parts and operating characteristics of the switched message network has been published. Refs. 2 through 11 describe some of the significant operating and engineering features.

The connections that are established in completing telephone calls show a very large variation in characteristics that are of importance to the transmission of data signals. This stems primarily from two factors:

i. There are a large variety of transmission systems used in the telephone plant (see Refs. 12 through 22). Table I lists some of the more important ones used in the Bell System today.

ii. The number of switched links (trunks) that are used to make up a given connection is quite variable. It is significant that a given long distance call may have as many as nine trunks switched in an over-all connection, or it may have as few as three. Two telephone calls between the same places may go over entirely different routes, pass through different offices and use different numbers of switched trunks.

The system characteristics usually of interest for data transmission include amplitude-frequency response, envelope delay-frequency characteristic, net loss, noise and echo suppressor turnaround time. In the case of voice-frequency cable, loaded or nonloaded, the characteristics are a function of length of loop or trunk. In a carrier system, the noise performance is a function of length and repeater spacing. As a practical matter, attenuation, envelope delay and noise also vary somewhat between channels of the same carrier system.

There are many cases where, for one reason or another, a particular trunk between switching offices is made up of a number of different transmission systems. The resultant trunk characteristics are then a combination of the characteristics of several systems.

In order to relieve the Data-Phone customer of the responsibility for engineering to accommodate the variability in transmission characteristics, subsets (usually modems with various control features) are provided. These act as buffers between customer data-generating or data-using equipment and telephone line characteristics, and provide well-defined interface arrangements.

III. DESCRIPTION OF THE FIELD MEASUREMENT PROGRAM

Data transmission characteristics of telephone connections have a wide range of variability. On the same basis, the error performance of Data-Phone systems might be expected to be quite variable and dependent on a large variety of conditions. An evaluation of data performance in the

TABLE I—BELL SYSTEM TRANSMISSION SYSTEMS USED ON MESSAGE TRUNKS

Type of Transmission System	Transmission Medium	Primary Application	Degree of Use	Remarks
Voice, open wire	Wire	Interlocal office trunks and class 5 to higher class offices	Small	Mostly rural application
Nonloaded voice cable	Cable	Interlocal office trunks and class 5 to higher class offices	Large	In most cases lengths are short (few thousand feet)
Loaded voice cable, all types[13]	Cable	Interlocal office trunks and class 5 to higher class offices	Large	
Type C carrier[14]	Open wire	Between higher class office	Medium	
Type H carrier[15]	Open wire	Class 5 to higher class offices and short haul between higher class offices	Small	
J carrier[16]	Open wire	Between higher class offices	Small	
M carrier	Open wire or power line	Class 5 to higher offices	Small	Also used for telephone loops in rural areas
K carrier[17]	Cable	Between higher class offices	Large	
L carrier[18]	Coaxial cable	Between higher class offices	Large	
N carrier[19]	Cable	Interlocal offices, class 5 to higher class offices and short haul between higher class offices	Large	
O carrier[20]	Open wire	Between higher class offices	Large	
ON carrier[21]	Cable	Interlocal office trunks, class 5 to higher class offices and short haul between higher class offices	Large	
TD system[22]	Radio	Between higher class offices	Large	
TJ system	Radio	Between higher class offices, short haul	Small	Large degree of use expected
TH system	Radio	Between higher class offices		Not yet in service — large degree of use expected

face of such variability leads inevitably to the use of sampling techniques and descriptions in terms of statistical distributions and probabilities. A field testing program designed to sample this variety of conditions and situations has been undertaken.

The objectives of the field testing program on the switched message network were to determine the following:

i. the statistics of error performance, permitting evaluation of error detection and error correction techniques where necessary;

ii. the factors that cause error to occur;

iii. the data speed capabilities and practical operating conditions, and the factors that limit them;

iv. the statistics concerning basic transmission characteristics — this is invaluable in designing new or improved data transmission systems.

Arrangements were made to place calls, send data signals and measure transmission parameters and error performance. Teams equipped with mobile testing terminals made telephone calls between varieties of locations throughout the country. The locations selected were places where potential Data-Phone customers were most likely to be found—in business districts, commercial areas and suburban industrial sites. Testing was carried out within, around and between New York, Chicago, Dallas, San Francisco and Los Angeles. These areas were selected as representative of the variety of conditions and facilities that exist in the present telephone network.

In the program about 1100 test calls were made. About 25 per cent of these were local calls not involved with the long distance switching plan. About 25 per cent were short-haul long distance calls, of up to about 400 miles airline distance. The remaining 50 per cent were long haul, 400 to 3000 miles long.

In order to keep the testing program within manageable size, a single data transmitting system known as the FM digital subset[23] was used for the higher-speed data performance tests. In this system the modulator accepts baseband binary information in serial form and provides a frequency-modulated output. The marking condition is one frequency, the spacing condition another. A single oscillator swings between the two frequencies and transmits the binary information to the demodulator. The demodulator is a zero-crossing detector, pulse generator and integrator, which provides serial binary baseband signals at its output. The output signals are reproductions of the modulator input signals modified by distortion effects of the telephone facility and modulation-demodulation process.

The error statistics of the telephone network that were determined

TABLE II—TRANSMISSION MEASUREMENTS

Type of Measurement	Conditions of Measurement	Measuring Equipment
Amplitude-frequency response	Between 600-ohm terminations at intervals of about 200 cps	Western Electric Co. 21A transmission measuring set
Envelope delay-frequency response	Between 600-ohm terminations at intervals of about 200 cps.	Acton Laboratories 451 & 452 envelope delay set
Steady noise	F1A weighting	Western Electric Co. 2B noise measuring set
Impulse noise — two methods	Counts in 30 minutes above given power levels, 144 weighting	Western Electric Co. 2B noise measuring set and General Radio 1556-A impact noise analyzer
	Counts in 30 minutes above given power levels, data system band filter weighting	Electronic slicer and counter
Noise recording	Unweighted 5-kc band	Ampex Model 307 magnetic tape recorder

with the FM modem reflect the characteristics of that data system. Measurements taken with some other type of modulation system would probably be somewhat different. In order to minimize the need for testing other types of systems under the conditions encountered during the tests, basic transmission characteristic measurements were also made on

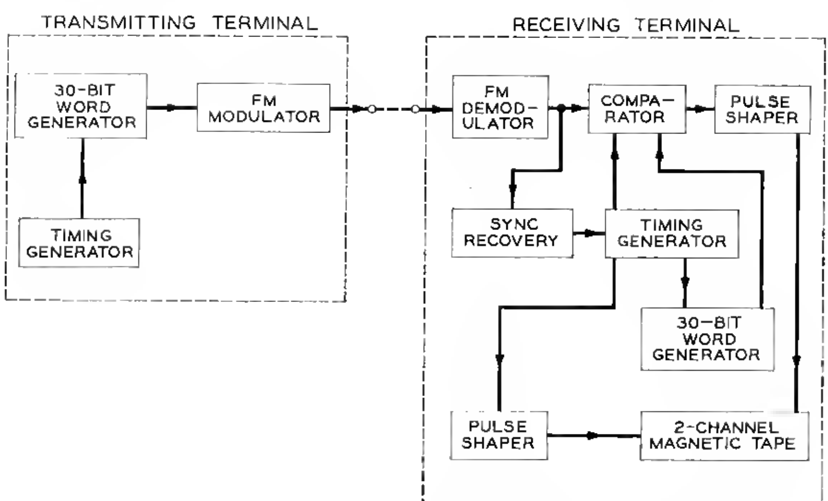


Fig. 1 — Block diagram of transmitting and receiving data terminals.

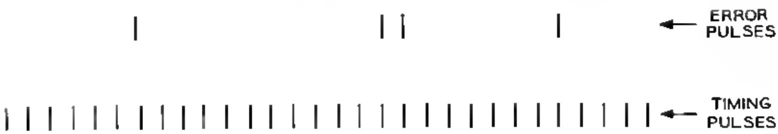


Fig. 2 — Method of error recording on magnetic tape.

each connection. At a later time these conditions may be simulated in the laboratory and comparisons made between the FM system and other modulation systems. A list of the more significant measurements is shown in Table II.

A block diagram showing the transmitting and receiving data terminals with associated circuitry is shown in Fig. 1. A 30-bit word generator, timed by a tuning fork clock,[24] was used to drive the FM subset modulator at the transmitting end. At the receiving end the line signal was demodulated and fed into a comparator. There the binary signal was compared bit by bit with a locally generated 30-bit word. A sync-deriving circuit provided timing information for the receiving terminal clock. The comparator circuit provided an output voltage spike whenever the demodulated 30-bit word and the locally generated 30-bit word did not compare on a bit-by-bit basis. This error indication was recorded on one track of a two-channel magnetic tape; the other channel recorded the locally generated timing signal. This method of recording the time and error information as bits in error and good bits between bits in error (see Fig. 2) permitted later analysis of error statistics by machine methods. The coding of the 30-bit word that was used for the error rate tests is shown in Fig. 3. This coding was selected to provide representative limiting conditions.

Additional circuitry and equipment were provided to permit varying the operating conditions of the data terminals. One of the limiting factors on data performance is the signal-to-noise ratio at the receiving end. Other things being constant, this is directly proportional to the signal level at the transmitting end. The transmitting level should, of course, be as high as is consistent with satisfactory operation on the telephone facility.

The maximum level permissible on telephone facilities is limited by two considerations:

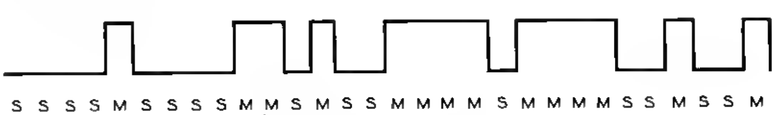


Fig. 3 — Coding of 30-bit word.

i. the coupling loss to other circuits operating in the same cable, open wire or carrier system;

ii. the power-handling capacity of carrier or radio system grouping amplifiers or modulators — overloading may cause modulation products to be generated that will fall in other channels of the same system.

For the tests, a level of -6 dbm at the transmitting terminal was selected to meet established criteria for interference and overloading. Means were provided to reduce the output level in discrete steps, so that relationships between error rate and transmitting level might be determined.

Based upon previous studies and experience, it was determined that the most satisfactory operating region was likely to be centered somewhere between 1200 and 1800 cps and that, at least initially, a speed of 600 bits per second should be used for these tests. Provision was made to operate the data terminals at three pairs of mark-space frequencies: 900(M)–1400(S), 1400(M)–1900(S) and 1100(M)–1900(S).

Information gathered during the early part of the program indicated that sufficient margin was available to permit increasing the speed if the effect of amplitude and delay distortion could be reduced. Compromise amplitude and delay equalizers were designed, and the latter part of the program was carried out using a speed of 1200 bits per second, with mark-space frequency pairs of 1100(M)–2100(S), 1200(M)–2200(S) and 1300(M)–2300(S).

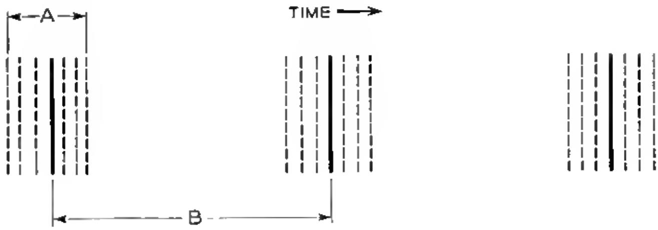
In order to accommodate the increased frequency spectrum, the digital subject was modified with a more optimum bandpass filter and integrating filter.

Error rate information was taken at 600 bits per second using the 900(M)–1400(S) frequency pair. At 1200 bits per second, the 1100(M)–2100(S) pair was used. The three frequency pairs at each speed were used to determine the best operating region for each connection. This was done by measuring maximum repetitive jitter in the transitions of the 30-bit word binary signal as received at the output of the demodulator. (Jitter is the total variation in time of the binary transitions from what they should be; the timing standard is supplied by the receiving clock.) The peak jitter may be expressed in terms of per cent peak distortion in accordance with the following (as shown on Fig. 4):

$$\text{per cent peak distortion} = \frac{\text{max. variation in transition time}}{\text{time of two bits}} \times 100.$$

(The maximum possible distortion is 50 per cent.)

The per cent peak distortion (repetitive jitter) was used as the criteria for determining the best pair of operating frequeucies.



PER CENT DISTORTION $= \frac{A}{2B} \times 100$

MAXIMUM POSSIBLE = 50 PER CENT

Fig. 4 — Measurement of peak jitter in terms of per cent peak distortion.

The following paragraphs discuss the effect of present telephone facility transmission characteristics on binary data signals and summarize the results of some of the basic transmission measurements that were made during the field testing program.

IV. BASIC TRANSMISSION CHARACTERISTICS

Considerable progress is being made in establishing the relationships between data system performance and the transmission parameters characterizing the telephone network. A general presentation is not within the scope of this paper and, indeed, would be premature at this time. However, the field sampling of such channel characteristics as amplitude and delay distortion, noise and net loss, together with a limited theoretical study, have made possible a first-order description of the data transmission capabilities of the telephone plant. The measured data are presented herein on a statistical basis.

4.1 *General*

Nyquist[25] theorizes that a channel should be capable of transmitting binary digital information at a rate numerically equal to twice the channel bandwidth, e.g., 6000 bits per second, assuming a bandwidth of 3000 cps. This requires a channel having flat loss and no delay distortion within the passband and infinite loss outside — conditions not met by the switched telephone network. Telephone bandwidths have been designed to accommodate speech frequencies from about 300 cps to about 3300 cps. It is therefore necessary to translate the data signal spectrum into this nominal passband by such means as the FM digital subset. If the resulting sidebands are transmitted symmetrically, the allowable bit speed is reduced by one-half.

Another factor limiting data speeds involves an effect of nonlinear distortion. It is frequently called "Kendall effect"[26] because its occurrence was first predicted by B. W. Kendall in connection with studies of telephoto transmission. Nonlinear distortion results in second-order modulation products that may fall within the baseband spectrum of the data signal. If this overlaps the line signal, distortion will result. Therefore the lower portions of tbe telepbone band cannot always be used efficiently and the frequency space available for the data signal is reduced.

Practically speaking, then, data speeds of binary signals on the switched telephone network are certainly less than 3000 bits per second, although higher speeds may be achieved by other than binary systems. For a given speed, the rate at which errors occur will depend on the method of modulation and transmission characteristics of the channel. The basic transmission phenomena of interest are:

i. *effective channel bandwidth*, characterized by the attenuation and delay distortion parameters of the telephone network, which imposes an upper bound on transmission speed and reduces the noise margin to error generation;

ii. *circuit net loss*, which affects signal-to-noise margins and hence margin to error;

iii. *noise*.

4.2 *Transmission Characteristics of the Telephone Plant*

The characteristics described herein represent the cumulative effects of the different transmission systems and switching equipment required to complete each particular connection. Consider initially the effects of individual transmission and switching facilities.

The attenuation of typical nonloaded wire pairs is proportional to the square root of frequency within the voice band and only for short lengths is this distortion across the band tolerable. Cable pairs longer than about three miles are loaded at uniform intervals with inductance to reduce attenuation frequency distortion and the over-all loss. With this added inductance, the line looks like a low-pass filter and exhibits a cutoff. Fig. 5 is a plot of the attenuation per mile for 22-gage pairs, both loaded and nonloaded, as a function of frequency normalized to the loaded pair cutoff frequency. The cutoff of the loaded facility also introduces additional phase or delay distortion over the nonloaded pair, as shown in Fig. 6.

Carrier systems exhibit cutoffs both at high and low frequencies, as shown in Fig. 7. For clarity, the reference flat loss values are displaced

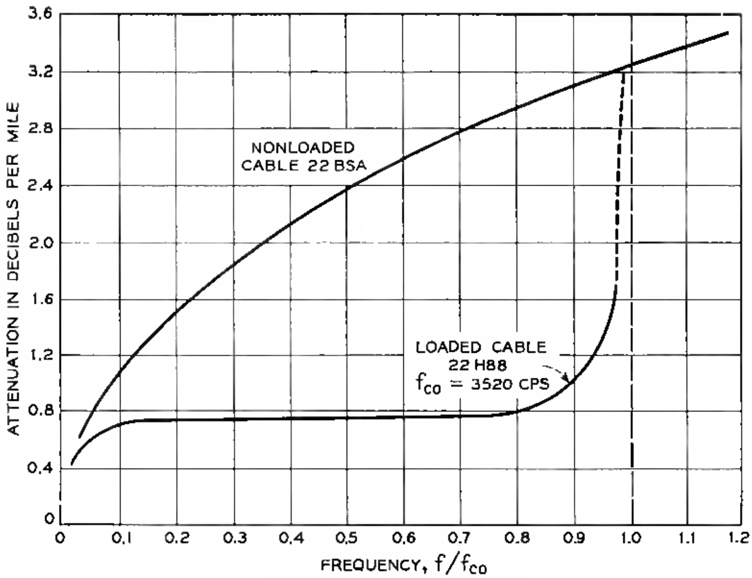


Fig. 5 — Attenuation characteristics, nonloaded and loaded cable.

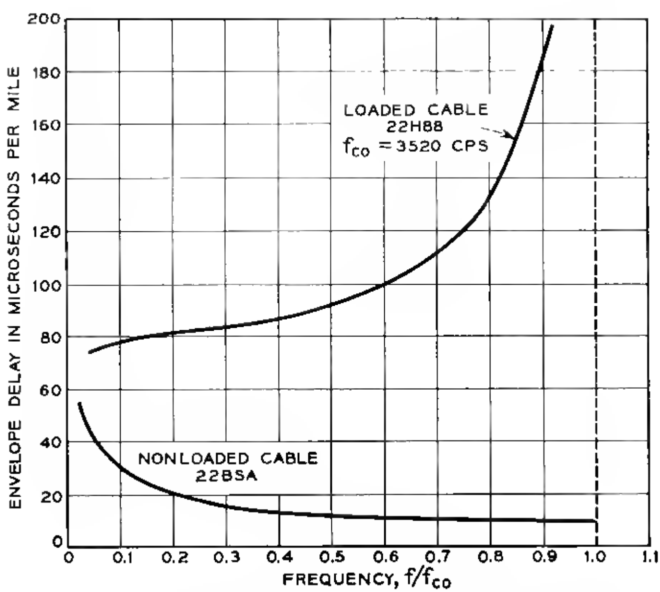


Fig. 6 — Envelope delay characteristics, nonloaded and loaded cable.

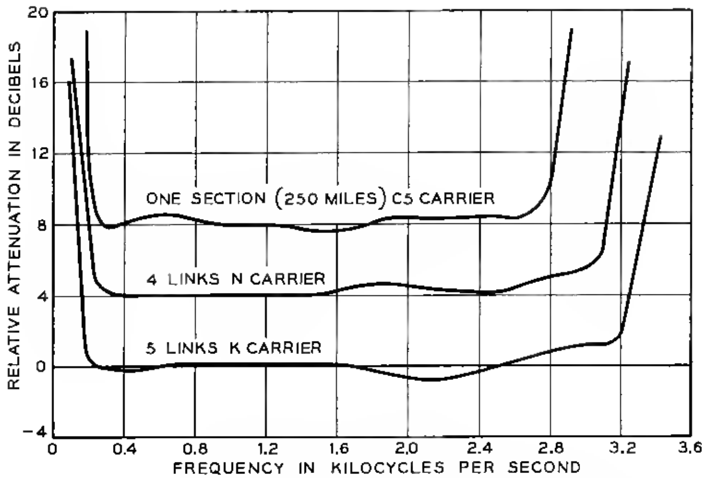


Fig. 7 — Representative attenuation characteristics, carrier systems.

vertically. Typical delay distortion characteristics for these systems are shown in Fig. 8.

Because of multiple connections and cabling runs within switching offices, shunt capacitance is added to a switched connection. This, of course, has the greatest effect at the upper end of the voice band on both attenuation and delay characteristics. Associated with switching points are the repeating coils, series capacitors and shunt inductors used for

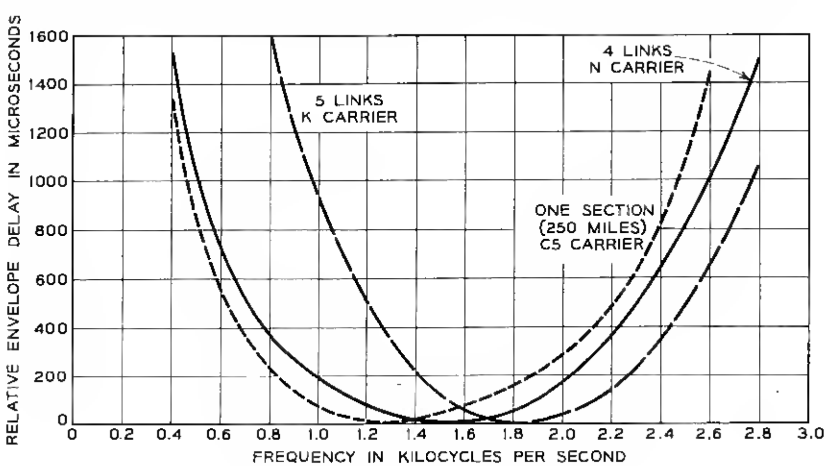


Fig. 8 — Relative envelope delay, carrier systems.

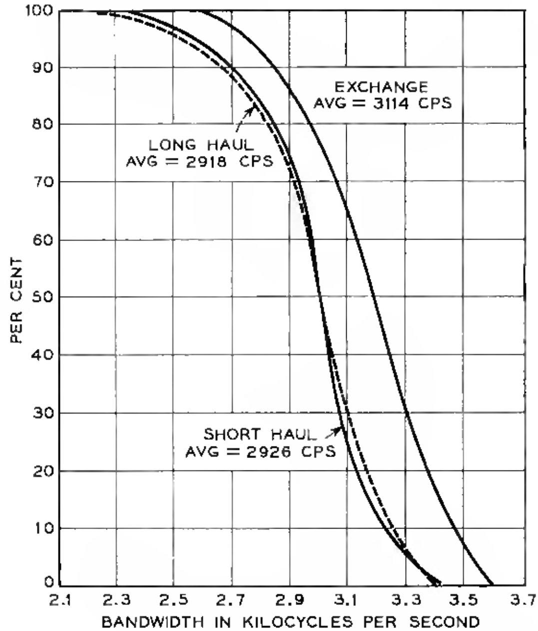


Fig. 9 — Cumulative distributions of 20-db bandwidths, showing percentage of circuits having bandwidths greater than that shown on abscissa.

signaling and supervisory purposes. These have their greatest effect at low frequencies. Therefore, even if the transmission facilities are voice-frequency wire lines, switched connections will show lower-end cutoffs.

At switching points a considerable amount of impulse noise is generated by the switches themselves, relays, dialing pulses, and the like. This noise is coupled in varying degrees, either directly or as cable crosstalk between pairs, to all channels switched by the office.

4.2.1 *Effective Channel Bandwidth*

It is convenient to consider attenuation and envelope delay distortions as occurring between two cutoff frequencies at which signal frequency components will be so severely attenuated by the transmission medium as to be relatively insignificant. For the purpose of this presentation, a 20-db bandwidth is defined as the interval between those frequencies at which the circuit loss is 20 db greater than the minimum loss of the circuit. Accordingly, Fig. 9 is a plot of cumulative distributions of 20-db bandwidths for the three classes of calls, showing the per-

centage of calls having bandwidths greater tban the corresponding abscissa value. Note that the average 20-db bandwidth is on the order of 3000 cps. However, distortions to be described within tbis band are such that considerably less than 3000 cps may be usable for data transmission purposes.

4.2.2 *Attenuation Distortion*

A careful examination of all the characteristics taken during the field measurement program has revealed that, in general, the relative attenuation characteristics assume the form shown in Fig. 10. That is, the circuit loss rises rapidly below f_1 cps and above f_3 cps, is relatively flat from f_1 cps to f_2 cps and rises linearly with frequency from f_2 cps to f_3 cps. These average frequencies and loss roll-offs are described in Table III.

For exchange calls, sharp lower roll-offs are not to be expected on the average, since many such connections are short, use voice facilities and cut off only because of the signaling and supervisory networks. Some longer calls use carrier facilities showing a sharper cutoff. Long distance calls, in particular, use single carrier systems extensively so that the average lower roll-off is fairly sharp.

The upper end roll-offs are much sharper for all classes of calls because of the combined effects of carrier systems and inductively loaded cable pairs.

Since data signals in most cases tend to be placed in the band from

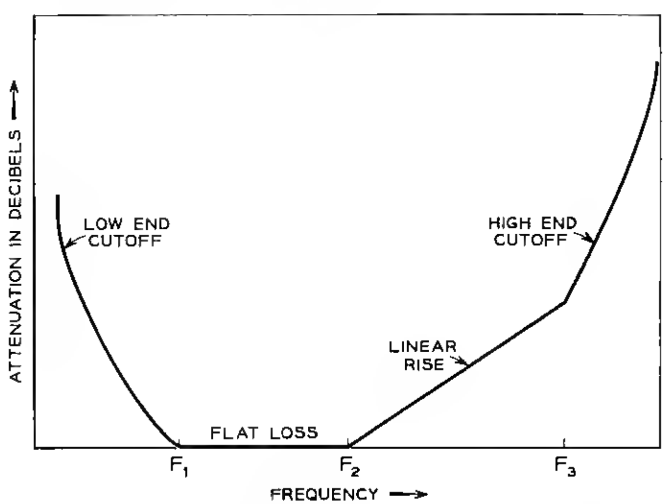


Fig. 10 — Relative attenuation characteristic of telephone circuits.

TABLE III

	Roll-Off Below f_1, db per octave	f_1, cps	f_2, cps	f_3, cps	Roll-Off Above f_3, db per octave
Exchange	15	240	1100	3000	80
Long distance					
Short-haul	24	300	1075	2950	90
Long-haul	27	280	1150	2850	80

1000 to 2600 cps, it is particularly desirable to describe the linear portion of the relative loss curve between these two frequencies. Fig. 11 is a plot of cumulative distributions for the loss differences between 1000 and 2600 cps for the three classes of calls. Note that, on the average, this difference is about 8 db but that, in about 5 per cent of the cases, 15 db is exceeded. In general, long distance connections show greater slopes, as a result, in part, of the shunt capacitance added by the switching points. Exchange calls usually are switched only twice, whereas long distance calls may be switched at four or more points.

With transmission at 1200 bits per second with the FM digital subset, it was found advantageous to use an attenuation equalizer designed to

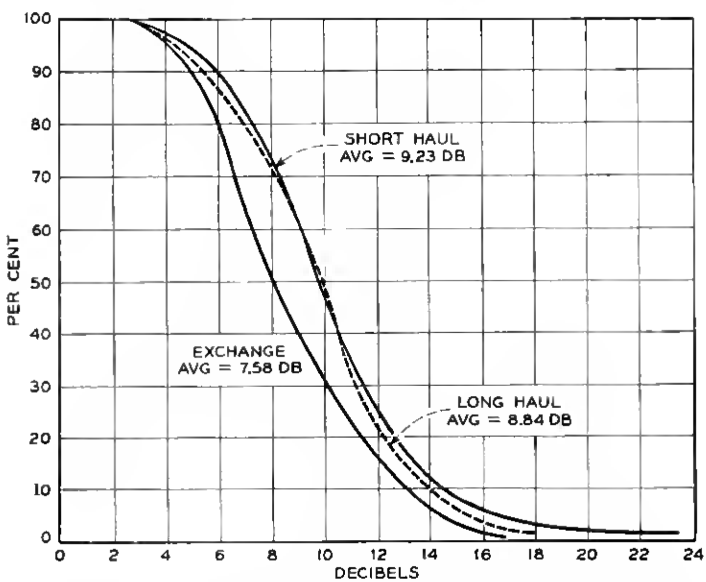


Fig. 11 — Difference in decibels between 1 kc and 2.6 kc — percentage of circuits having decibel difference greater than that shown on abscissa.

compensate for a 4-db slope between 1100 and 2100 cps, which are the mark and space data frequencies respectively. Between 1000 and 2600 cps this network equalized a loss slope of about 7 db.

4.2.3 *Envelope Delay Distortion*

The ear is relatively insensitive to minor phase distortions, so that the telephone message plant, designed for speech transmission, has not required the extremely low distortions demanded by data transmission. Since there is no reason to assume that the telephone network is minimum phase, knowledge of attenuation characteristics must be supplemented by a characterization of the phase variations. It is most practical to measure the derivative of phase with respect to frequency, $d(\theta)/d\omega$, which has the dimension of time and is referred to as *envelope delay*.

Curves of envelope delay versus frequency tend to be concave upward as a result of the upper and lower cutoffs of the telephone network. Average envelope delay characteristics are plotted in Fig. 12 for the three classes of calls, with the minimum envelope delays normalized to zero. They were derived by drawing smooth curves through the following five points: the average frequency of minimum delay (FMD), the average upper and lower frequencies at which the envelope delay is 1.0 millisecond greater than the minimum, and the average upper and lower 0.5-millisecond frequencies.

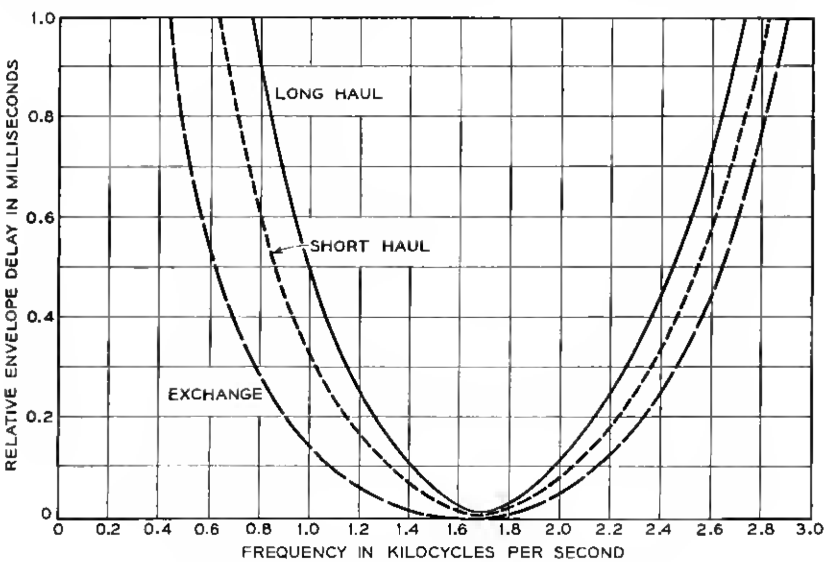


Fig. 12 — Average envelope delay characteristics.

Three facts are noteworthy: (a) the average FMD is on the order of 1700 cps; (b) distortion for exchange calls is less than for long distance calls and (c) all the curves appear to be fairly symmetrical about their respective FMD's.

It is mildly surprising that the exchange delay characteristics do not show more dissymmetry around an FMD somewhat lower than measured. Such a result is to be expected if the lower cutoff is determined by signaling and supervisory circuits. The explanation lies in the fact that almost 50 per cent of the exchange connections measured used carrier facilities with typically symmetrical delay curves. Deleting the data from calls using exchange carrier systems gives rise to the curve shown on Fig. 13, which is somewhat more representative of wire line characteristics.

Of interest are the variations from these average curves that are shown in Fig. 14. For each point used to draw the average characteristics described above, limits were found so that about 90 per cent of the measured points fell within these limits. By systematically joining respective limit points, the plots in Fig. 14 were derived. Careful sampling of the actual data confirms that approximately 90 per cent of the measured curves do fall within the shaded areas of the diagram, even though the

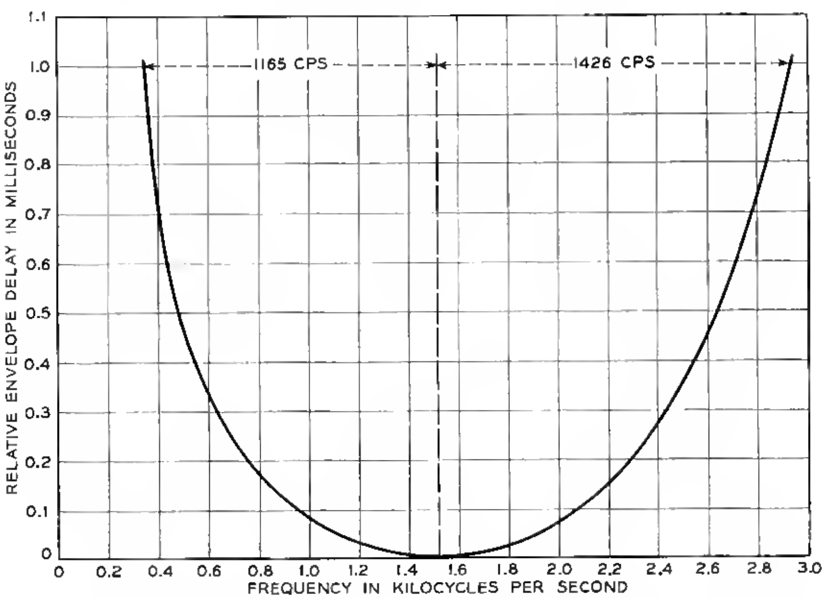


Fig. 13 — Average envelope delay characteristics for exchange calls on cable.

limit points were determined independently. Dotted lines indicate typical measured curves.

Note that all curves are tangent to the abscissa representing minimum delay (zero microseconds) at frequencies varying from 1200 to 2000 cps. More detailed information on the variations of this frequency of minimum delay is shown in Fig. 15.

Statistics on the delay distortion at the band edges are presented in Figs. 16 and 17 in terms of 0.5-millisecond and 1.0-millisecond "bandwidths." Delay bandwidth is here defined as the difference between those frequencies at which the envelope delay distortion is 0.5 or 1.0 millisecond.

A comparison is made in Fig. 18 of the measured delay characteristics with the compromise delay equalizer used during the tests with trans-

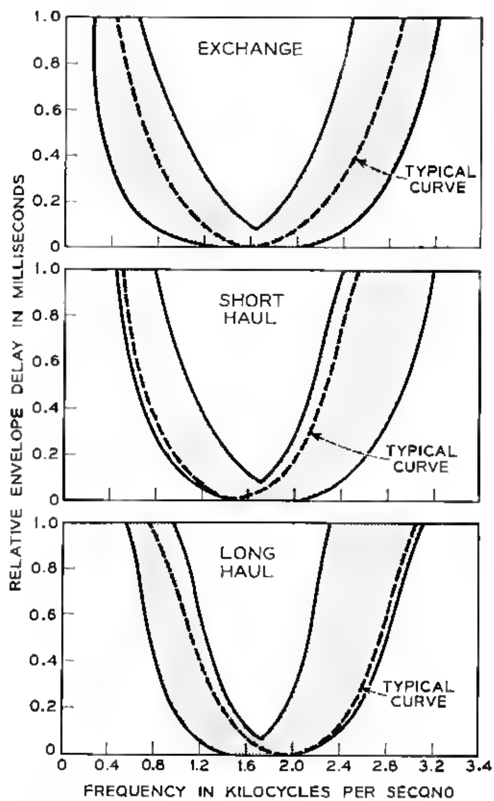


Fig. 14 — Envelope delay distortion — locus of 90 per cent of circuit characteristics.

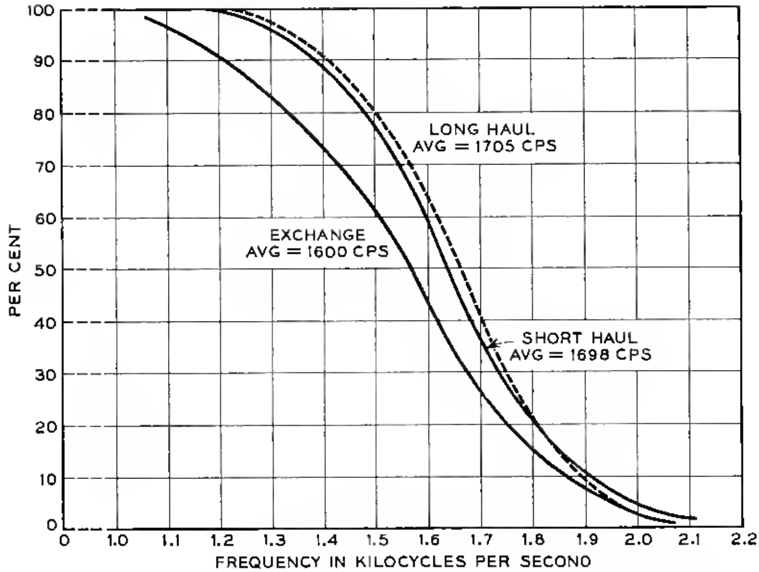


Fig. 15 — Frequency of minimum envelope delay — percentage of circuits having frequency of minimum delay greater than that shown on abscissa.

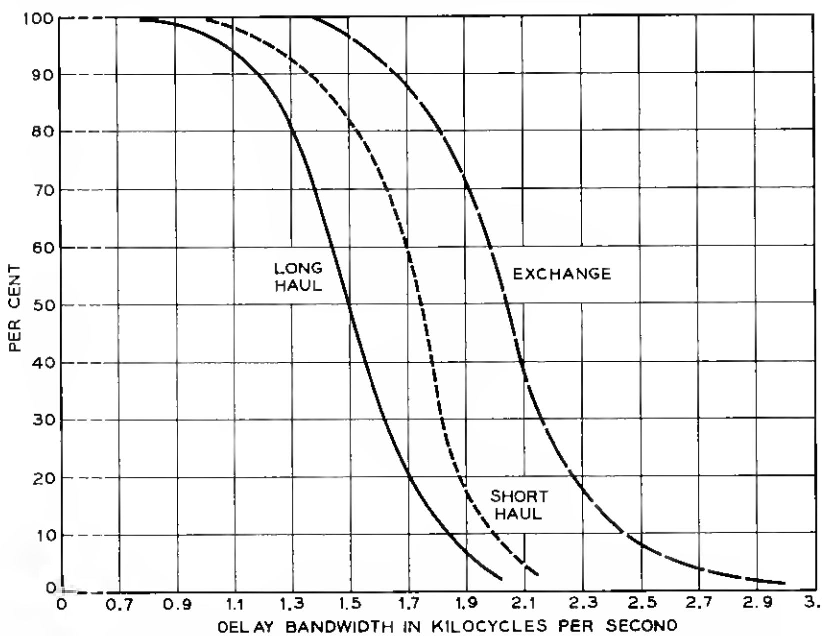


Fig. 16 — Percentage of circuits with 0.5-millisecond delay bandwidth greater than that shown on abscissa.

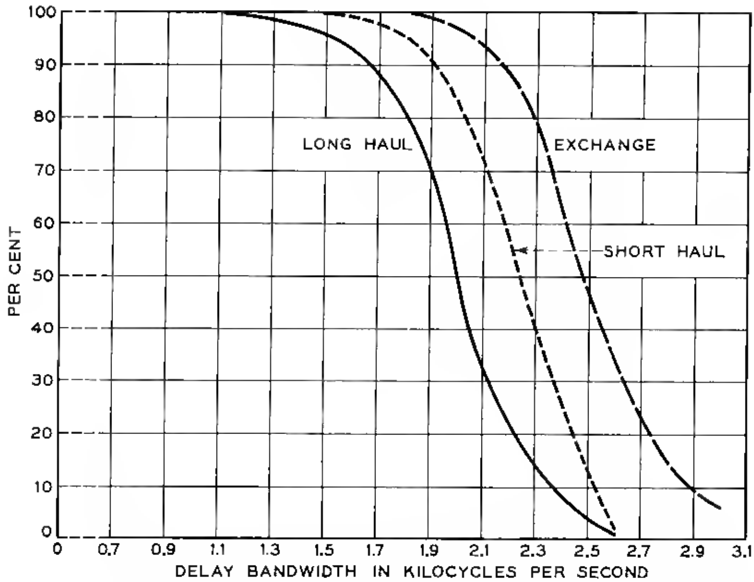


Fig. 17 — Percentage of circuits with 1-millisecond delay bandwidth greater than that shown on abscissa.

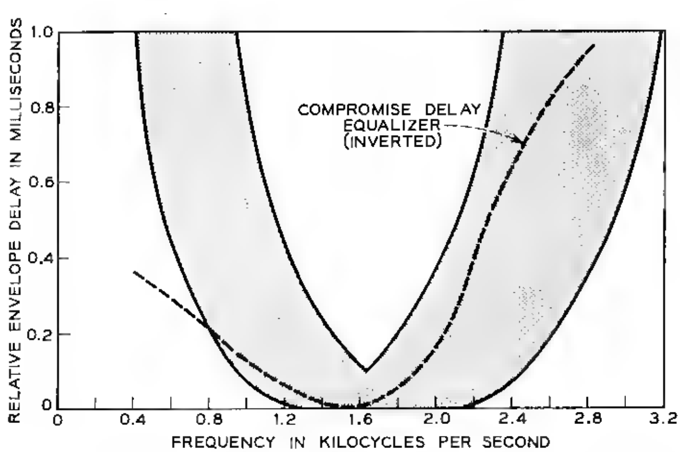


Fig. 18 — Envelope delay distortion characteristic for all calls — locus of 90 per cent of all circuit characteristics.

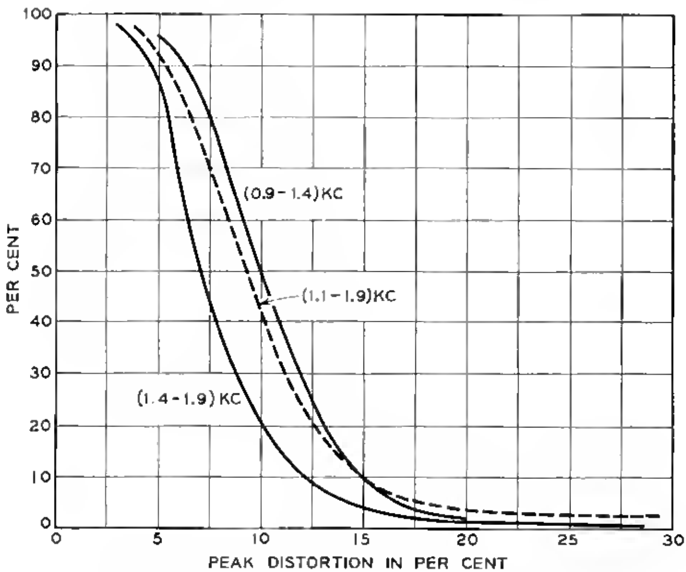


Fig. 19 — Peak distortion for 600-bit-per-second calls, no equalization — percentage of circuits having peak distortion greater than that on abscissa.

mission at 1200 bits. The inverse of the equalizer characteristic is shown superposed on a diagram indicating the locus of 90 per cent of the delay curves including all three classes of calls. Hindsight indicates that lower frequencies probably would have been better equalized on the average had the compromise favored those circuits utilizing carrier facilities.

The combined effect of amplitude and delay distortion on the FM digital subset shows up as jitter on the transitions of the demodulated signal and can be described in terms of peak distortion (repetitive jitter). Peak distortions of less than 20 per cent are considered quite acceptable. Fig. 19 shows that more than 99 per cent of the calls met this 20 per cent limit while transmitting at 600 bits at mark-space frequencies of 1400–1900 cps. Although the percentage of calls exceeding the limit did not vary widely for the three frequency pairs used, the over-all distribution for the 1400–1900 cps was considerably better. This was probably due to a better match of the resultant data spectrum to the average envelope delay characteristic. A correlation of peak distortion and error performance showed that the error statistics would not have been significantly changed if the 1400–1900 cps frequency pair had been used in this test.

Upon changing to a speed of 1200 bits per second, the measured peak

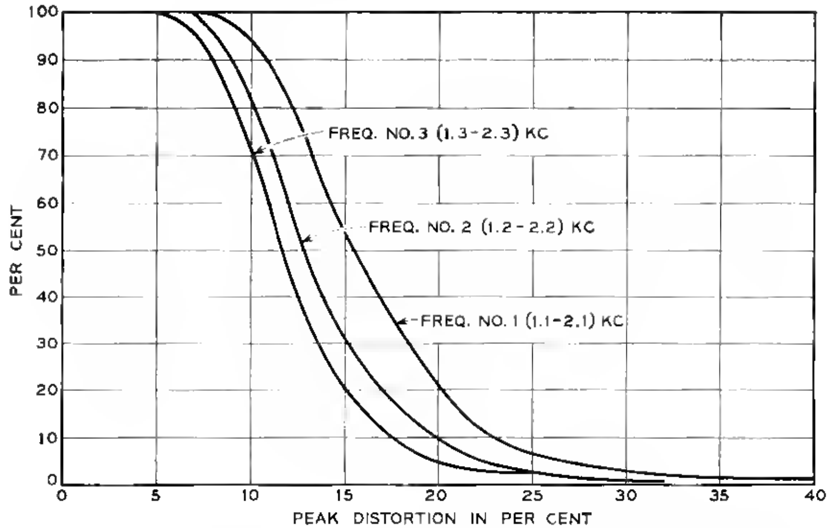


Fig. 20 — Peak distortion for 1200-bit-per-second calls with equalization (delay plus attenuation) — percentage of circuits having peak distortion greater than that shown on abscissa.

distortion, without either attenuation or delay equalization, was beyond practical limits. After the compromise equalizer was inserted, the jitter was considerably improved. Fig. 20 plots the cumulative distributions of peak distortion for the three pairs of frequencies used. Note that, for pair #1 (1100–2100 cps), over 20 per cent of all calls exceeded the 20 per cent limit, whereas for pairs #2 (1200–2200 cps) and #3 (1300–2300 cps) less than 10 per cent of the calls exceeded the limit. Pair #3 shows the lowest distortion for two possible reasons: (a) these frequencies best match the average compromise equalized connection and (b) the number of carrier cycles per signal element is greatest for this pair.

4.2.4 *Ripples in Distortion Characteristics*

The attenuation and delay characteristics described herein were derived by discounting ripples in the raw measured data. These ripples, mainly the result of echos (reflected energy), can be appreciable and must not be ignored. The source of echoes is varied and includes all points of impedance mismatch in bilateral circuits, and hybrid unbalance at the junction of two- and four-wire circuits. If multiple echoes exist, ripples in the attenuation and delay characteristics of the same circuit are not necessarily correlated. However, for each characteristic the ripple period on a frequency scale tends to be inversely proportional to the

electrical length of the path from the observer to the source of echo. That is, close-in echoes give rise to long sweeping ripples, while remote echoes cause fine structure ripples.

Due to increased reflected energy at the band edges, where impedances deteriorate, the ripple amplitudes tend to increase in these areas. In the main, the ripple in the amplitude characteristic is significant (i.e., greater than one to two db) only above about 2000 cps. An appreciation of the amount of ripple likely to be encountered can be gained by referring to Fig. 21, where a bar chart indicates the percentage of circuits having a maximum peak-to-peak ripple in decibels. For the most part, this maximum ripple occurs in the frequency range of 2000 to 3000 cps.

Both the amplitude and period of the ripples vary across the band, probably due to the existence of multiple echo points. Such variations are difficult if not impossible to describe statistically.

It has been pointed out[27] that close-in echoes result in ripples in transmission characteristics that can be equalized, whereas remote echoes cannot. The reason for this is that an individual transmitted pulse will be affected mainly by its own echo if the source of the echo is close in, and can be equalized to eliminate this distortion. Remote echoes tend to affect subsequent transmitted pulses, and the effect is random for an information bearing train. Hence ripple equalization will not be effective in general.

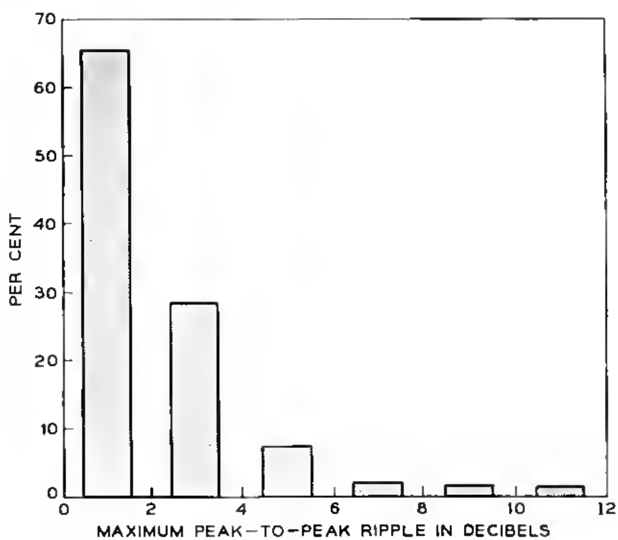


Fig. 21 — Percentage of circuits with maximum peak-to-peak amplitude ripple.

4.2.5 *Circuit Net Loss*

It is common practice to specify the net loss of telepbone circuits at 1000 cps, but the aetual loss for a complex frequeney signal may be somewhat different, depending on the attenuation frequency characteristic of the circuit. An example of this difference will be given. Consider first the cumulative distribution of 1000-cps circuit net loss (CNL) for the three classes of calls in Fig. 22. Note that, on the average, excbange calls are a few decibels better than long distance calls. This is to be expected, since loss tends to be a function of the physical length of the connections. Since long distance connections can be thousands of miles longer than exchange calls, it is gratifying to note that this relatively small difference in loss hetween the two types has been achieved in practice.

Consider the loss experienced by the FM digital subset signal operating at 1200 bits per second. For mark-space frequeneies of 1100–2100 cps tbe apparent carrier frequency is 1600 cps. Reference to Section 4.2.2 shows that the average loss at 1600 cps is about 3 db above the 1000-cps value. Taking into account the entire average attenuation characteristic across this data band — 500 to 2700 cps — an excess loss

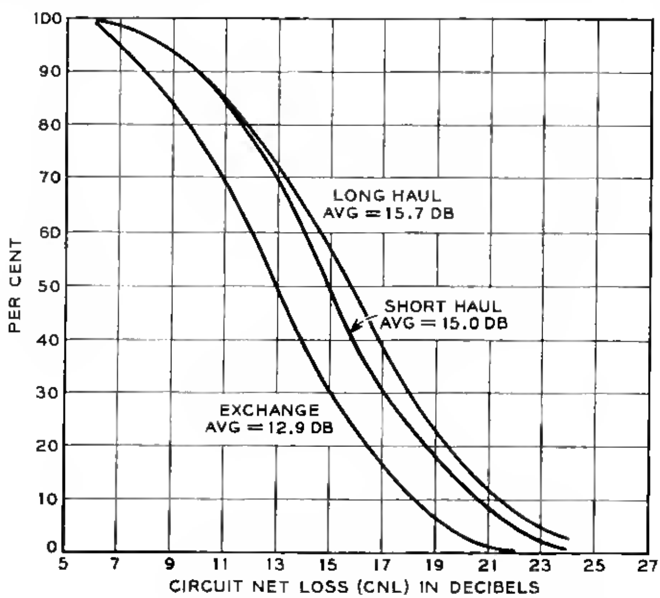


Fig. 22 — Percentage of circuits with 1000 cps net loss greater than that shown on abscissa.

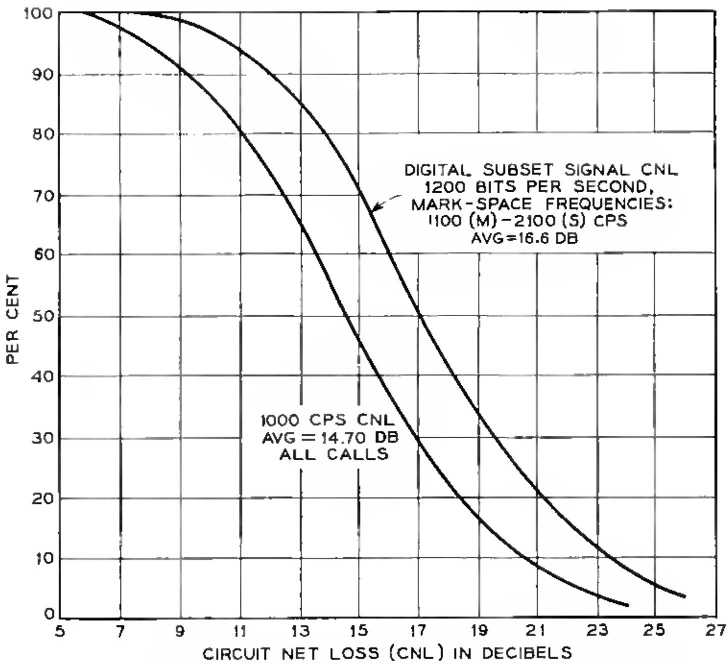


Fig. 23 — Percentage of circuits with circuit net losses greater than that shown on abscissa.

of less than 3 db over 1000-cps CNL is to be expected. Fig. 23 is a plot in the cumulative sense of the 1000-cps CNL and the data signal net loss for all calls. Note that the average data signal loss is only 2 db greater than the average 1000-cps CNL, confirming the prediction.

4.2.6 *Noise*

Two types of noise are of interest in the telephone plant: (a) steady line noise and (b) impulse noise. Steady noise is important for its interfering effect on speech transmission. Impulse noise, characterized by relatively high peaks of short duration pulses, has the greatest effect on the transmission of pulses.

To see that, in general, steady noise is not of great importance in pulse transmission, consider its cumulative distribution in Fig. 24. Note that only about 1 per cent of all the calls exceed noise values of 40 dba. This is equivalent to an average of about −42 dbm of white noise in a 3-kc band. Referring again to Fig. 23, note that in only about 5 per cent of the calls did the −6-dbm data signal exceed losses of 26 db for a

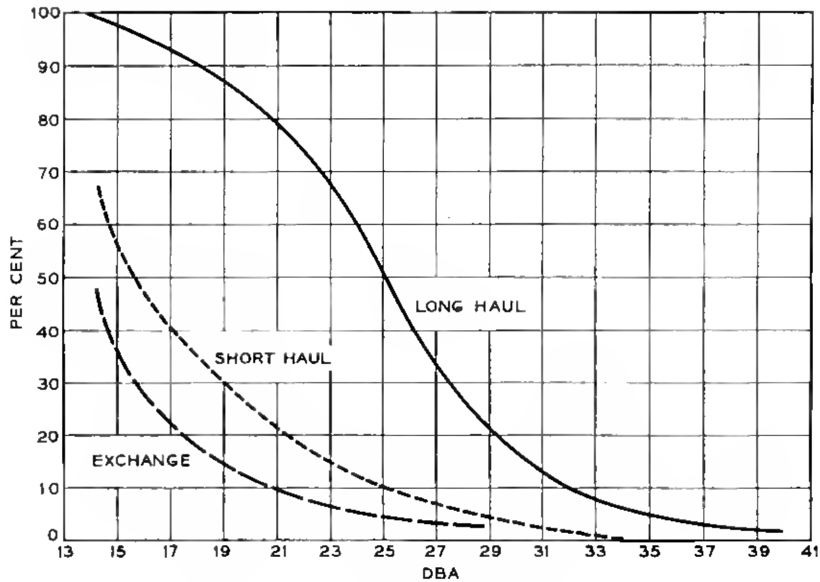


Fig. 24 — Percentage of circuits with F1A noise greater than that shown on abscissa.

received level of −32 dbm. Even without determining the degree of interdependence of the two distributions, it is apparent that very few calls exhibited less than a 10-db average signal-to-noise ratio.

Impulse noise, on the other hand, frequently has peaks comparable to the received data signal level. The incidence of impulse noise tends to follow the traffic fluctuations in the switched network. That is, busy periods generate considerably more impulse bits than do quiet periods. In fact, in the field test about 40 per cent of the calls failed to show any impulse counts regardless of the measured level. This is to be expected, since calls were placed at random during both the busy and quiet periods of the day.

The average number of counts of impulse noise above given slice levels for 15-minute measurement periods is plotted in Fig. 25. These data give a general indication of impulse noise conditions within the message plant even though they do not correlate well with error rates on the same calls. In many instances in which errors occurred, no impulse noise was measured, and vice versa. As a result, the correlation of impulse noise and error generation was poor. Drop-outs and interruptions that do not show up as impulse noise counts limit the usefulness of noise measure-

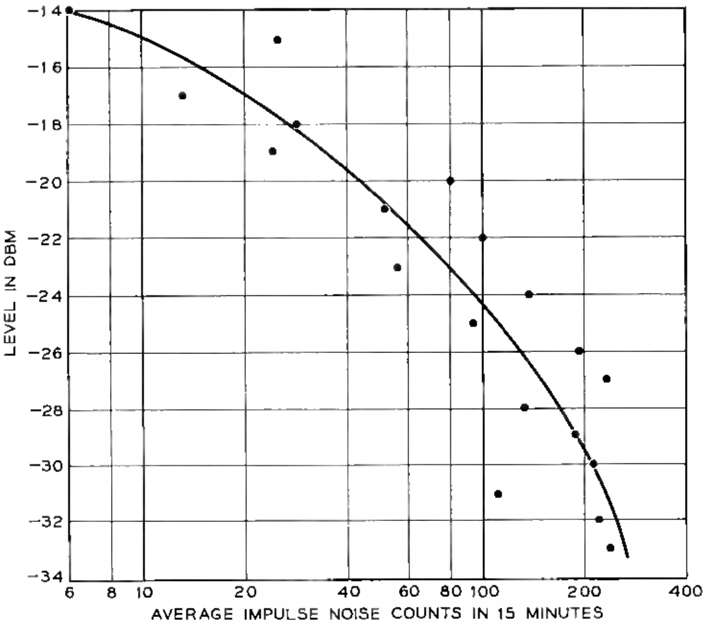


Fig. 25 — Average impulse noise counts in 15 minutes above level shown on ordinate.

ments in predicting error performance on the switched message network.

The error performance actually experienced during the field testing program is described and discussed in the following paragraphs.

V. ERROR RATES

The primary purpose of the investigation described herein was to provide statistics of the error rate and the time distribution of errors at bit rates that have a *high probability of success for transmission over any facility between any two telephone stations in the United States.* There would be little value in designing data systems that had such high bit rates that they would work on only half of the circuits encountered. We are interested in the performance at data rates where satisfactory transmission can be achieved on practically all of the connections.

Success in data communication does not mean that the communication must be completely error-free. The terms "successful" or "satisfactory" are as difficult to define for data as they are for speech com-

munication. Very few people carry out a conversation — even in the same room, much less over a communication facility — without the necessity of repetition or the deliberate insertion of redundancy in vital parts of the message because of distraction on the part of the listener. This distraction is often extraneous noise or reverberation in the room. On a communication facility, this distraction might be noise on the circuit or distortion of the signal resulting in nearly the same effect as noise on the circuit. Therefore, a request is made on the part of the listener to repeat. The communication is usually considered successful or satisfactory until tbe point is reacbed where the distraction becomes high enough to require an annoying amount of requests for repetition. This will vary with the articulation and modulation of the speaker, the text of the message and the patience of tbe communicators, as well as tbe transmission characteristics of tbe circuit. For Data-Phone equipment the same philosophy applies. The evaluation of performance is more easily defined in data because of the binary nature of the information and because the so-called distraction results in a recognizable error. However, redundancy either in the form of repetition or check digits, or both, can be used to improve tbe accuracy, and the need for it is a function of tbe same variables. By tbe proper use of redundancy it is possible to achieve any desired degree of accuracy.

In order to obtain a better understanding of error rate as a function of transmitting level, some measurements were made at a number of levels. The bulk of the data, which includes the distribution of errors as a function of time, was collected witb a transmission level of -6 dbm at the sending station.

The metbod of recording clock pulses and error pulses on magnetic tape, as previously described, permits a computer to count the number of good bits between error bits and present the distribution of errors. This distribution is obtained in a printed output similar to that shown below, and is also available on cards and on magnetic tape, wbicb can tben be used for later evaluation of various types of error-control schemes or for more detailed analysis of error bursts:

Zeros	*Ones*
25,226	1
222,866	1
14,692	6
8,971	1

The first column designated "zeros" is the number of good bits between crrors. The second column designated "ones" is the number of

errors. Thus, this printed output is interpreted as follows: 25,226 good bits were transmitted and then one error was encountered; then 222,866 good bits were transmitted and another error was encountered, then 14,692 good bits were transmitted and six consecutive errors were encountered, etc. This is the basic information from which various types of analysis have been made.

The particular distributions and relations presented herein have been selected on the basis of what is thought to be most significant in the planning of data communication systems. The first statistic essential in the planning or evaluation of a data system is the cumulative distribution of average error rates.

5.1 *Average Error Rates at 600 Bits Per Second*

Figs. 26 through 29 indicate the probability of obtaining a circuit that produces an average error rate better than that shown on the abscissas. These probabilities are shown for the three types of calls. It is at the receiving central office where the introduction of switching noise is most critical due to the lower level of the signal. Fig. 26 indicates that 85 per cent of the exchange calls can handle 600 bits per second with an error rate of one bit in error for every 10^5 bits or more transmitted. A slightly lower percentage, 82 per cent, of the short-haul calls performed as well, and 75 per cent of the long-haul circuits met this accuracy figure. On the

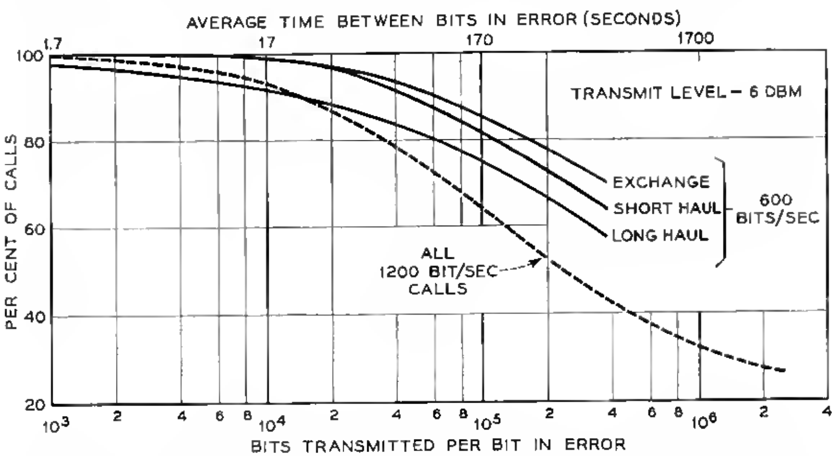


Fig. 26 — Error-rate distribution by class of call, 600 bits per second — percentage of calls with average error rate equal to or better than that shown on abscissa.

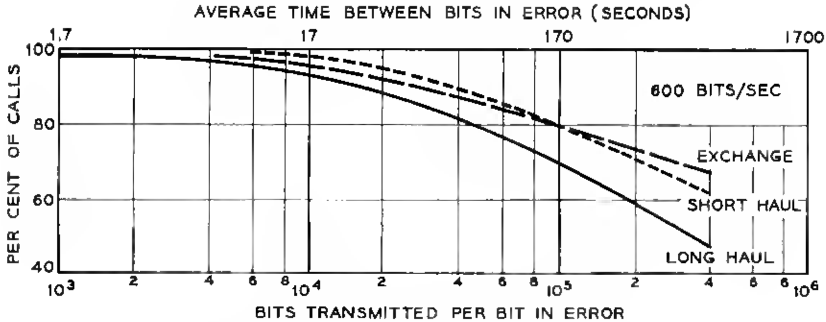


Fig. 27 — Percentage of calls having average error rate better than that shown on abscissa — crossbar office receiving, receive level −25 db.

average, exchange calls have less attenuation than short-haul or long-haul calls. Since all stations are transmitting at the same levels, this means that the signal-to-noise ratio at the receiving station is greater for the exchange calls.

In order to eliminate the effect of the higher losses on the longer connections, Fig. 27 compares error rates at a common receive level of −25 dbm, the transmitting levels being adjusted so that signals of −25 dbm were received at the receiving station line. Here the exchange calls and the short-haul calls are virtually the same, but on the long-haul calls at

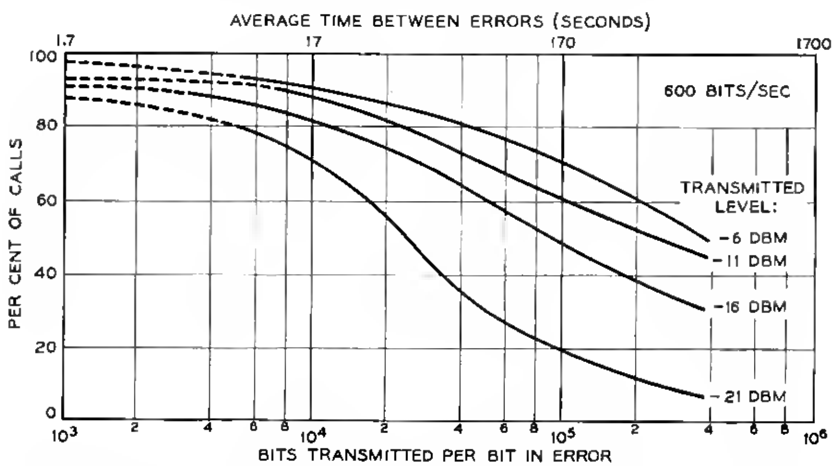


Fig. 28 — Long-haul toll calls — percentage of circuits with error rates better than that shown on abscissa as transmitting level is reduced.

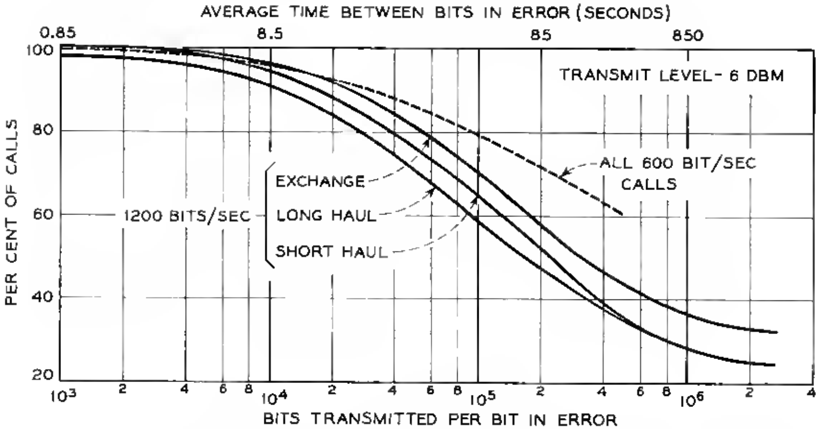


Fig. 29 — Error-rate distribution by class of call, 1200 bits per second — percentage of calls with average error rate better than that shown on abscissa.

the same receive level the performance is somewhat inferor. This should be expected, since on long circuits there is a greater probability of exposure to noise and interference.

Fig. 28 indicates the change in error rate distribution as the transmitting level is reduced in 5 db steps. Note that if the transmitting level is lowered from −6 dbm to −11 dbm the error rate is approximately doubled.

5.2 *Average Error Rate at 1200 Bits Per Second*

The curves in Fig. 29 are plots of the results at 1200 bits per second for the various types of calls. They are shown on the same axis with the curve of all calls at 600 bits per second. Note that, for a given percentage of the circuits, the error rate at 1200 bits per second is two or three times greater than that for 600 bits per second. In other words, 70 per cent of the calls at 600 bits per second will produce an error rate better than about one error in 200,000 bits, but at 1200 bits per second 70 per cent of the calls will produce an error rate better than one in 70,000 bits.

VI. ERROR DISTRIBUTIONS AND ERROR-CONTROL EVALUATION

A data transmission system should be designed to provide for the optimum useful bits of information with the minimum cost. Cost includes the cost of data equipment such as the data originating and receiving equipment and the cost of providing a communication channel,

as well as that of the modulators and the demodulators. In many cases, this resolves into the design for optimum line efficiency for a specified accuracy objective. Many studies have been made to relate this efficiency in terms of various error correcting and error detecting methods. Wood[28] derives optimum block lengths for retransmission methods, and Brown and Meyers[29] describe the efficiency of various error-control systems, including forward-acting codes and retransmission methods. However, in all these evaluations the probability of errors and the distribution of errors in time are fundamental in arriving at the proper solution. The selection of optimum codes and optimum block lengths in error-control schemes is a complex subject. The information contained in the statistics herein is provided to aid in the derivation of better control systems. However, the over-all system concept for data transmission, including error control, should be cognizant of the following considerations:

i. How serious is an error that is produced? Is error control necessary in view of the accuracy of the origin of the data or the final disposition of the data?

ii. Is the relationship of the line transmission cost to equipment cost including error control such that optimum line efficiency may not result in the most economical solution for the system?

iii. Is the format of the data such that optimum blocking must be in terms of lines of characters or numbers of cards where mechanical limitations are an important factor in the optimum arrangement?

iv. Is there storage and logic circuitry already provided in the system, such as in a computer or in buffer storage of other data machines, which can also be used for error control purposes?

The above factors are functions of the data machinery and how it is employed. In addition, the functions of the transmission medium, such as error probability, propagation time and turn-around time of echo suppressors, must be considered to resolve the optimum data transmission system. If it were not necessary to consider all these factors, then the error-control function could become a basic feature of the transmission medium. Therefore, it is not the purpose of this paper to make an evaluation of the many specific error-control methods that have been proposed, but it is desired to provide the fundamental error distributions and indicate the relative orders of magnitude of improvement that might be expected from the error-control schemes. The following curves are arranged to facilitate evaluation of optimum block lengths for retransmission methods, to evaluate error detection schemes, and to evaluate forward-acting single-error and multiple-error correction codes, including burst-correcting codes.

The error-rate distribution curves (Figs. 26 through 29) describe the probability of getting an error per number of bits transmitted. An important statistic is the probability of getting succeeding errors within various time intervals after the first error, for it is the dependency of one error on another that must be considered in error-detection or error-correction codes. If a cumulative distribution is made of the numbers shown in the previous table under the "zero" column, which represents the good bits between errors, a curve is obtained which shows the probability of getting an error as a function of time since the previous error. Figs. 30 and 31 show these distributions for 600 bits per second and 1200 bits per second, respectively. The results have been analyzed and plotted for exchange calls, short-haul calls and long-haul calls, and another curve for all calls together has been drawn.

These curves provide statistics that are useful in the planning and evaluation of error-control schemes. For example, after an error has occurred, the probability of having one or more good bits following that error before getting another error is 0.70 considering all types of calls. This means that the probability of having zero good bits, which is the

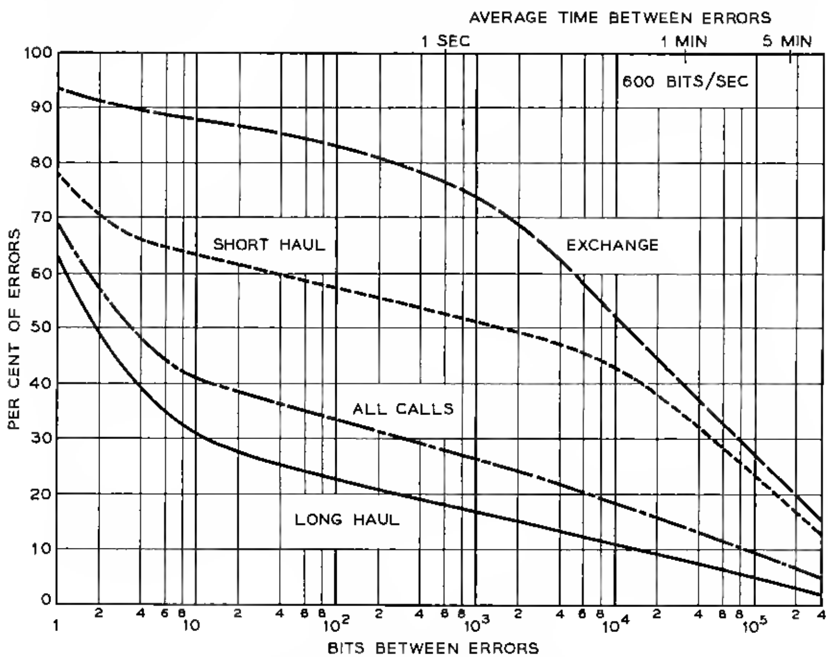


Fig. 30 — Error-free transmission time between successive errors, 600 bits per second — percentage of errors having as many as or more error-free bits between them as that shown on abscissa.

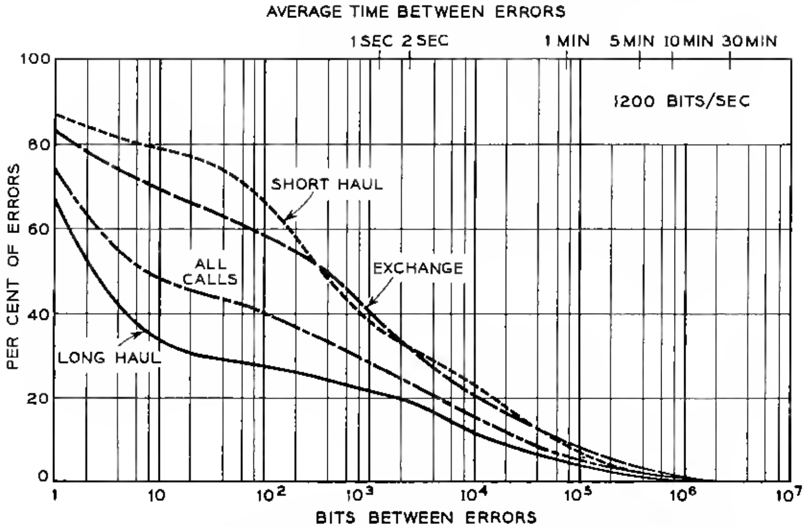


Fig. 31 — Error-free transmission time between successive errors, 1200 bits per second — percentage of errors having as many as or more error-free bits between them as that shown on abscissa.

same as having two or more consecutive errors, is 0.30. In other words, 30 per cent of all errors are immediately followed by one or more errors. For 1200 bits per second the comparable figure is 0.74, which is approximately the same as the 0.70 for 600 bits per second. If there is particular interest in eight-bit characters, for example, on long-distance calls at 600 bits per second, the curve shows that approximately 40 per cent of the erroneous characters are likely to contain single bit errors because four or more good bits will follow the erroneous bit. This assumes that, on the average, the erroneous bit is in the middle of the character. However, this means that 60 per cent of the erroneous characters will have more than one bit in error.

Each forward-acting correction code, whether it be a Hamming code,[30] a Hagelbarger code,[31] or a square matrix code, is limited in the number of errors it can correct within a given number of total bits. Also, some codes require that a period of error-free transmission exist for specific lengths of time between errors or bursts of errors in order to clear out the memory and logic of the circuitry to have it ready for the next burst of errors. The number of correctable errors in a burst and the clear-out period required is a function of the redundancy of the code and the amount of storage and logic provided in the system.

To define these bursts let us assume the sequence of good bits and error bits shown by zeros and ones below:

sequence: 00001010000000000110101000000001000100000
bursts of four: |↔| |↔||↔| |↔||↔|

A *burst* is defined as a sequence of bits that starts with an error bit and extends for $N - 1$ additional bits whether they be error bits or not, where N is the length of the burst. For example, assume we are interested in burst sizes of length 4. The first bit in error and the next three bits following are considered as the burst. The succeeding burst of size 4 starts at the next error that occurs after the first burst, and so on until the entire message is analyzed by bursts of size 4 and the quantity of good bits between bursts. Thus, in the illustration above there are nine good bits between the first two bursts of 4, one good bit between the second and third burst, six good bits between the third and fourth burst, and three good bits between the fourth and fifth burst. The number of good bits between bursts, as illustrated, is counted from the last error in one burst to the beginning of the next burst. The density of the burst is the ratio of good bits in a burst to total bits. For example, in the illustration two out of a total of four bits in the first burst are in error, and the density is 50 per cent, whereas in the second burst three bits are in error, and the density is 25 per cent.

An analysis of the 600-bit-per-second transmission and the 1200-bit-per-second transmission on this basis is described in Figs. 32 through 39. A range of burst sizes from bursts of one, which facilitate the evaluation

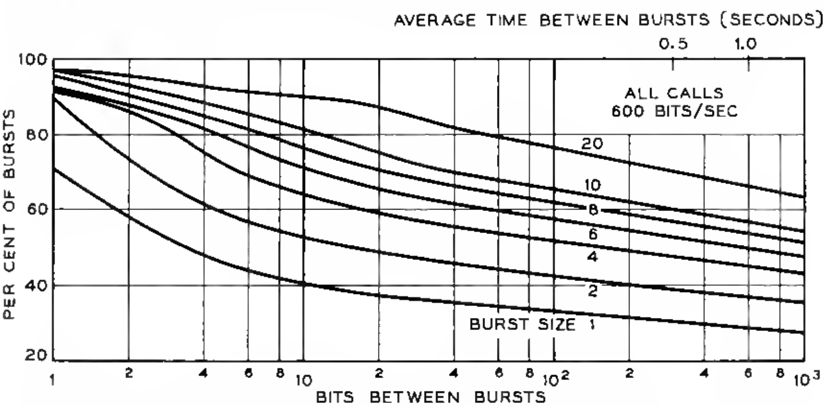


Fig. 32 — Error-free transmission time between successive bursts of various sizes, all calls, 600 bits per second — percentage of bursts having as many as or more error-free bits between them as that shown on abscissa.

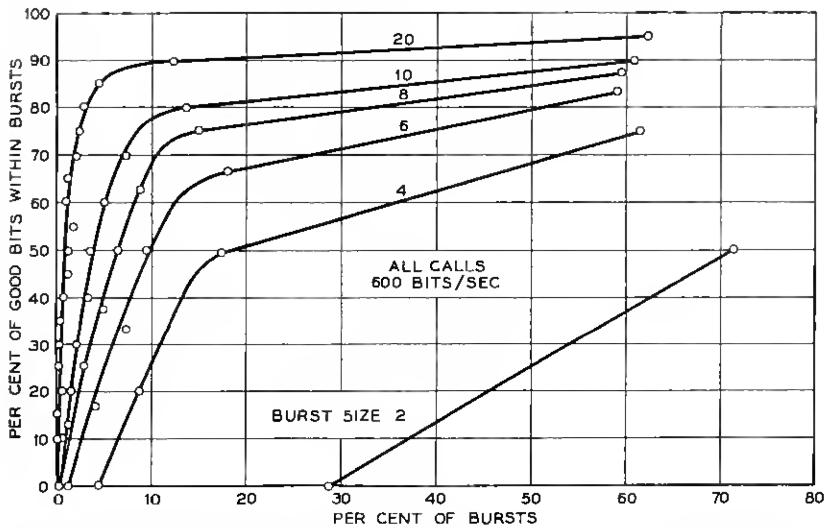


Fig. 33 — Density of good bits within bursts, all calls, 600 bits per second — percentage of good bits within bursts plotted as a function of the percentage of bursts of various sizes.

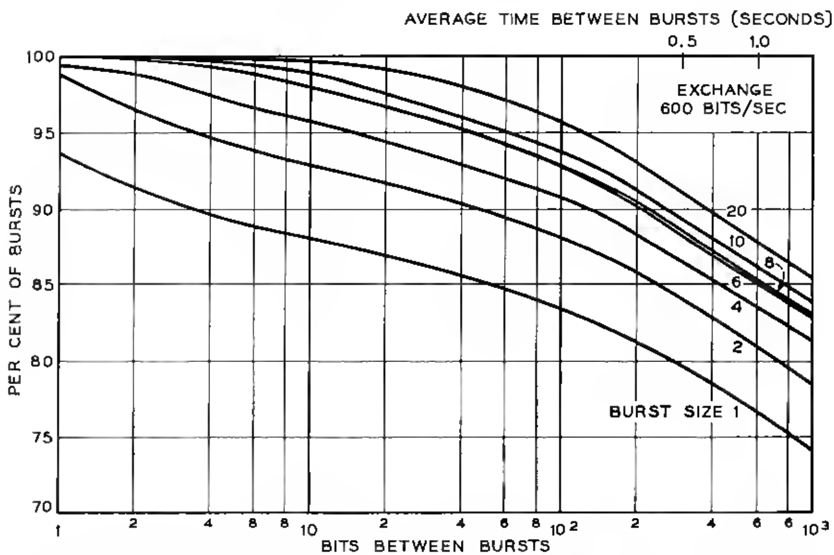


Fig. 34 — Error-free transmission time between successive bursts of various sizes, exchange calls only, 600 bits per second — percentage of bursts having as many as or more error-free bits between them as that shown on abscissa.

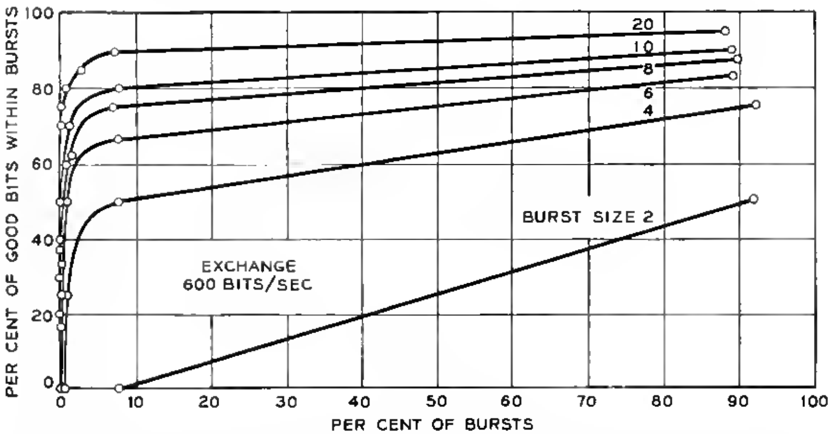


Fig. 35 — Density of good bits within bursts, exchange calls only, 1200 bits per second — percentage of good bits within bursts plotted as a function of the percentage of bursts of various sizes.

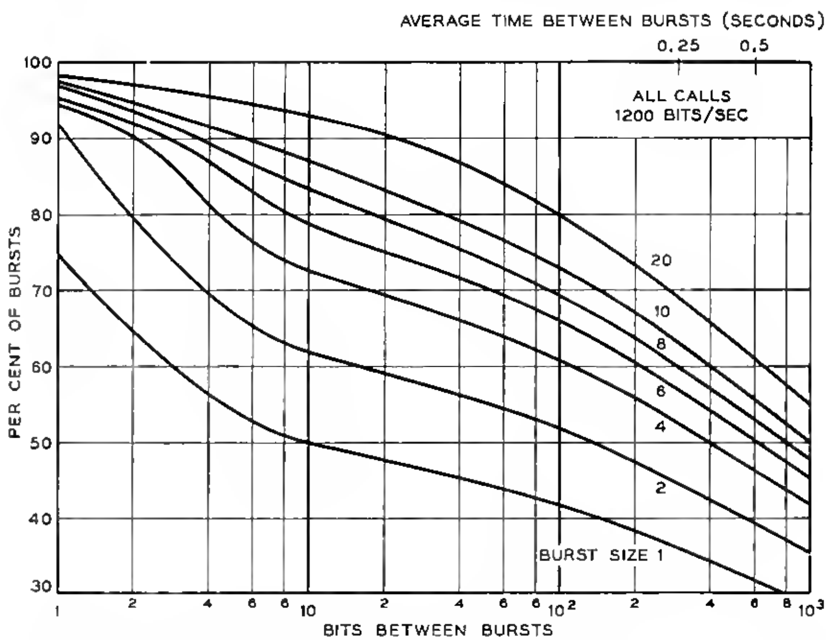


Fig. 36 — Error-free transmission time between successive burst of various sizes, all calls, 1200 bits per second — percentage of bursts having as many as or more error-free bits between them as that shown on abscissa.

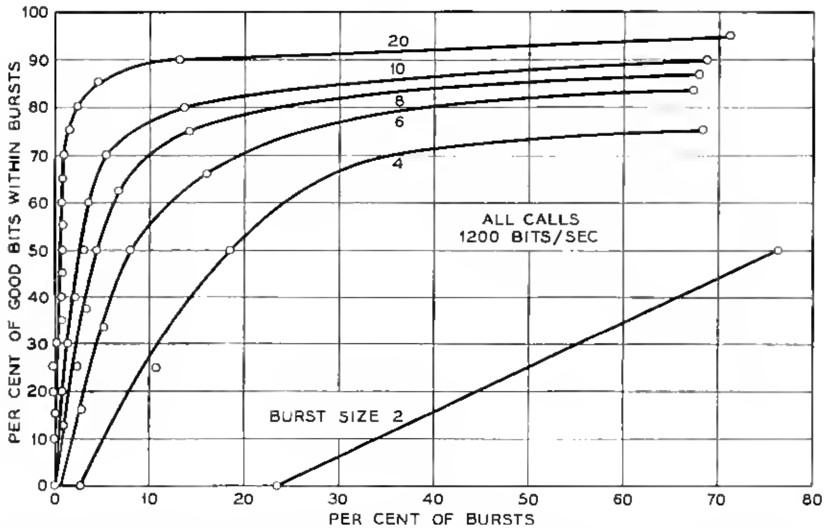


Fig. 37 — Density of good bits within bursts, all calls, 1200 bits per second — percentage of good bits within bursts plotted as a function of the percentage of bursts of various sizes.

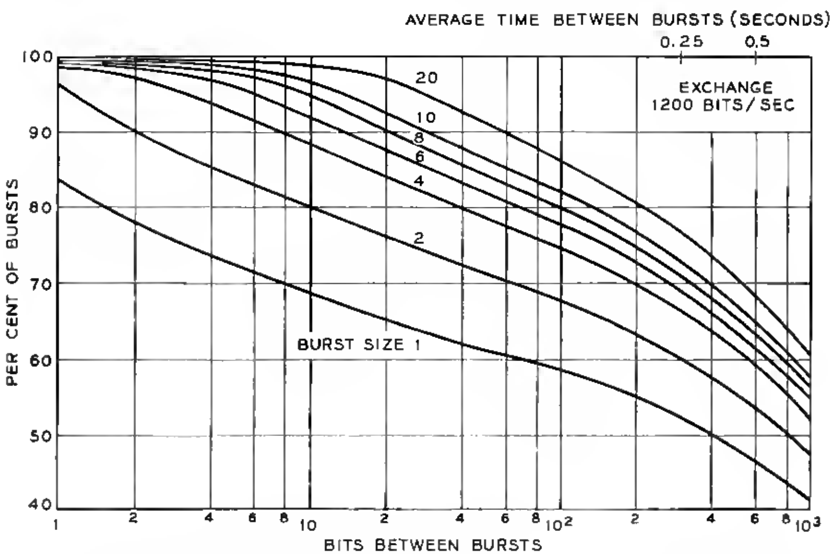


Fig. 38 — Error-free transmission time between successive bursts of various sizes, exchange calls only, 1200 bits per second — percentage of bursts having as many as or more error-free bits between them as that shown on abscissa.

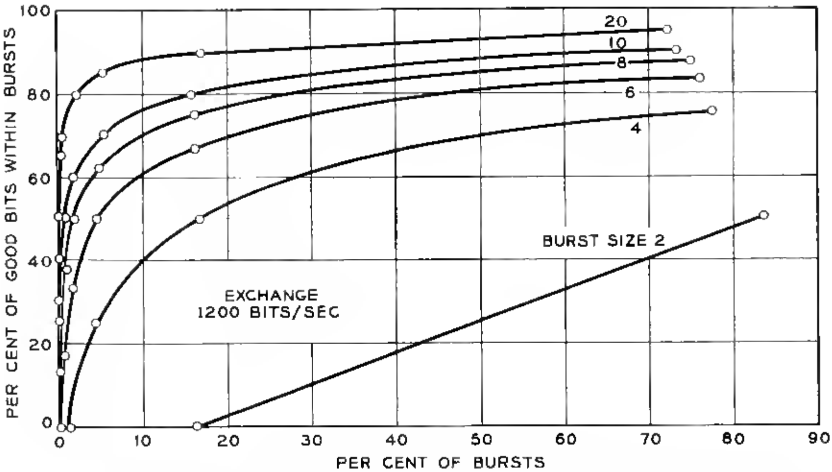


Fig. 39 — Density of good bits within bursts, exchange calls only, 1200 bits per second — percentage of good bits within bursts plotted as a function of the percentage of bursts of various sizes.

of single-error correction codes, to bursts of 20 is shown. This range is provided because it is felt that burst-correcting codes for larger bursts than 20 become quite complex and would be of such high cost that there would be little application for such systems. The curves are shown for all calls and for only the exchange calls, showing how the effectiveness of error-correcting schemes may vary for different types of calls.

To illustrate the improvement that can be expected from a Hagelbarger code, which is designed for correcting bursts of eight bits in duration and which requires a clear-out interval of 26 bits between bursts, an approximation is made. Such a code would have a redundancy of 50 per cent. In Fig. 32, 69 per cent of the bursts for burst length 8 have 26 or more good bits between them. This means these are correctible bursts. If it is assumed that the uncorrected bursts have the same error density as the corrected bursts — namely, that shown by Fig. 33 — then an improvement or reduction in average error rate of about 3.2 to 1 can be expected. The only reason why this is an approximation rather than an exact evaluation is because of the previously stated assumption regarding density of uncorrectible bursts, and also because the coding scheme may introduce additional errors when the bursts are too close. For exchange calls, based on the information shown in Fig. 34, approximately 96.5 per cent of the bursts are correctible by an eight-bit burst-correcting code with 50 per cent redundancy, which should result in an improve-

ment of about 28 to 1. This is because on exchange calls there are fewer uncorrectible bursts, since there are fewer bursts that extend beyond eight bits and fewer bursts that are closer together tban 26 bits. It is interesting to note that, if an evaluation is made of a single-error-correcting code that requires, say, 10 good bits between errors (a Hamming code would accomplish this), then it is found that on these excbange calls the single-bit errors predominate. Thus, a substantial amount of the improvement made with an eigbt-bit burst-correcting code could have been made with a single-error-correcting code.

Now we shall examine all the calls to explore the amount of improvement that may be expected by increasing the error-correcting capabilities from 8 to 20 bits. This means that for the same redundancy tbe clear-out interval must be extended from 26 to 62 bits. The curves indicate that tbere is very little additional improvement. Fig. 32 shows that tbe 20-bit bursts with a clear-out interval of 62 bits represent 79 per cent of the total burst instead of 69 per cent. Therefore, little advantage is obtained compared to the increased circuit complexity that must be provided. These bursts may seem long for data transmission, since they may effect many bits, but for speech the circuits are very satisfactory and such interruptions are rarely noticeable.

Information is provided for determining the effectiveness of these codes for different types of calls. However, the relative value of these coding schemes can better be illustrated on cumulative-error-rate distribution curves similar to those previously described in Figs. 26 and 29. A computer was programmed to correct those errors that were single errors with more than 10 good bits between them, and also was programmed to correct tbose bursts that did not exceed 8 bits in duration and had at least 26 good bits between them. These values were chosen since they are thought to represent coding systems tbat can be implemented with relative ease and illustrate order of magnitude improvements that might be expected. The cumulative distributions of uncorrectible errors are shown in Figs. 40 and 41 for speeds of 600 and 1200 bits per second, respectively. Also, plotted on the same axes are the identical curves previously shown in Figs. 26 and 29, which are distributions for these calls without error correction of any type. Thus, it is shown tbat at 600 bits per second 80 per cent of the circuits achieve an error rate better than one error in more than 100,000 bits, without any error correction. With single-error correction, 85 per cent of tbe circuits perform this well, and with burst correction the percentage is increased to 91 per cent. It is necessary to keep in mind that, witb error correction, redundancy is added and, in the case of burst correction, 50 per cent of

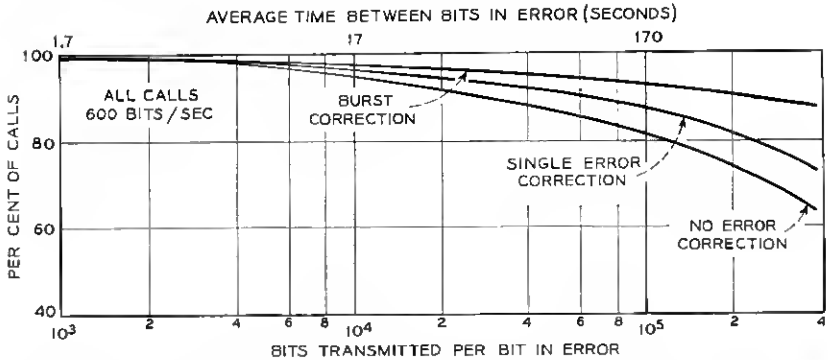


Fig. 40 — Error rate distribution, all calls, 600 bits per second — percentage of calls having an average error rate better than that shown on abscissa.

the bits are check bits. Therefore, a comparison is made of one in 10^5 bits with no correction, to one in 1.7×10^5 bits with single-error correction, and one in 2×10^5 bits with the burst correction.

At 1200 bits per second the improvement in performance with single-error correction and burst correction is somewhat better than at 600 bits per second. Actually, the addition of error control tends to make the performance at 600 and 1200 bits per second very close. For example, at 600 bits per second with burst correction, 94 per cent of the circuits produce an error rate better than one bit per 50,000 transmitted. At

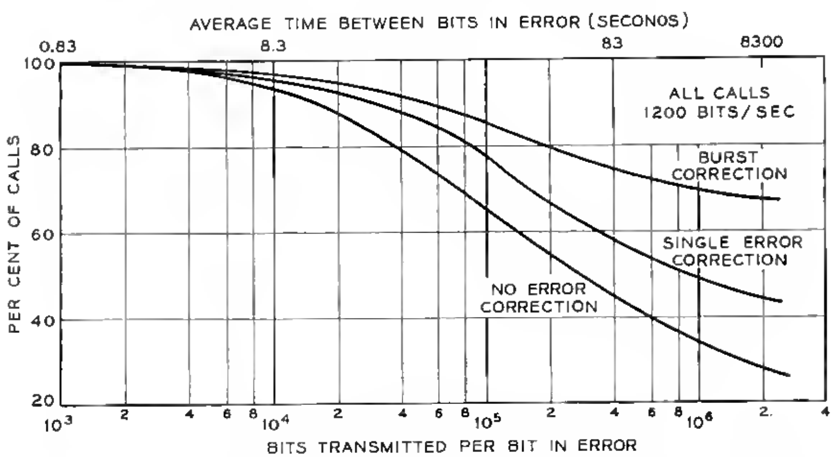


Fig. 41 — Error-rate distribution, all calls, 1200 bits per second — percentage of calls having an average error rate better than that shown on abscissa.

1200 bits per second with the same error control, 90 per cent of the circuits gave the same performance. The sets of curves in Figs. 40 and 41 are for all calls made in the investigation, and therefore include exchange, short-haul and long-haul connections. It is emphasized that if such curves were shown for only exchange calls the improvement would be greater, whereas if they were shown for only long-haul calls the improvement would be less.

These error statistics indicate that, where a high degree of accuracy is required, retransmission of data is also required. Forward-acting error-correcting codes by themselves do not at present appear to be the complete solution. Undoubtedly, progress will be made in the direction of achieving large volumes of storage at low cost, which will facilitate more economical design of forward-acting error-control schemes. Also, as new transmission systems are developed and improvements are made to existing systems, the probability of large bursts of errors will be reduced.

The previous curves have provided the information necessary to aid in making decisions as to whether error control is necessary and what type is most effective. If retransmission is necessary the question then arises as to the optimum block length. An important factor adding to the complexity of this problem is the turnaround of the echo suppressors, the propagation time, and the physical and electrical design of the data input and output machinery. Retransmission methods can cover a vast range of possibilities. For example, one method might be to send blocks of data of just a few bits in duration three times consecutively and take the best two out of three. Another scheme, which might represent the opposite extreme, would be to transmit entire messages, say 10 minutes in duration, and when an error is encountered retransmit the whole message over again. To evaluate the effectiveness of these schemes it is necessary to know the probability of error-free transmission as a function of message length. Also, if this latter scheme were used with single-error correction, so that retransmission would not be required on the single errors but only on the long bursts, this method of error control might be considerably more promising.

Figs. 42 and 43 describe the probability of error-free transmission for no-error correction, single-error correction and eight-bit burst correction. Because of the vast time scales that may be of interest for error-control purposes, the curves are plotted on two different scales. The probability scale on the left permits accurate evaluation for message lengths of less than 1000 bits and the scale on the right is used for longer message lengths. These curves are shown for both 600-bit-per-second and 1200-bit-per-second tests.

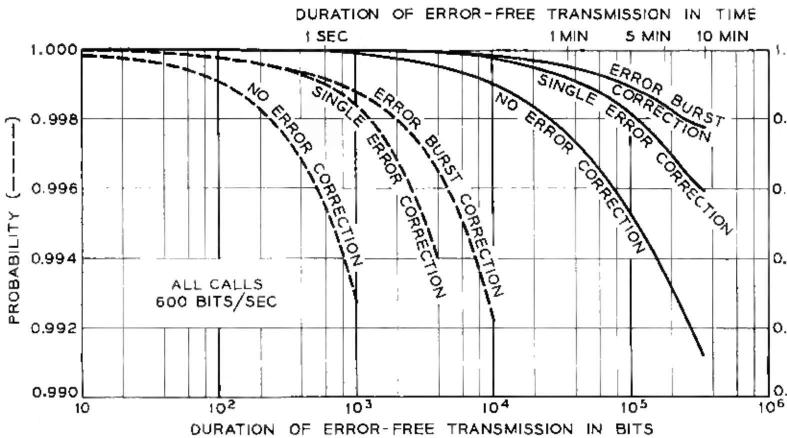


Fig. 42 — Probability of error-free transmission, all calls, 600 bits per second — probability of transmitting as many as or more error-free bits than that shown on abscissa.

For example, Fig. 42 indicates that, with 1000-bit blocks at 600 bits per second with no error correction the probability of error-free transmission is 0.993. With single-error correction this probability increases to 0.9984, and with burst correction this probability further increases to 0.9988. Thus, it is quite obvious that in this application there is very little advantage in forward-acting error-correction codes. A forward-

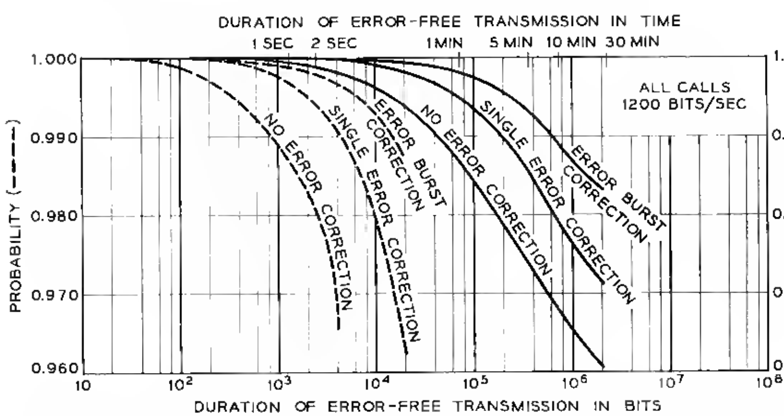


Fig. 43 — Probability of error-free transmission, all calls, 1200 bits per second — probability of transmitting as many as or more error-free bits than that shown on abscissa.

acting error-control scheme may not by itself appear very promising when it provides for a reduction in error rate by a factor, say, of 5 to 1. But if this error-correction scheme is used along with a retransmission method and the forward-acting code reduces the number of retransmissions, then this code may have proved itself by increasing the efficiency in the use of the telephone circuit. Many more interesting examples of error control could be discussed on the basis of these curves, but the objective herein is to illustrate the engineering value of the statistics and let individual ingenuity go to work.

VII. CONCLUSION

The evaluation program has demonstrated that speeds as high as 1200 bits per second with an FM modem using a zero-crossing detection system are entirely practicable on the regular switched telephone network. The error performance on the connections is variable, depending upon a number of factors. In many cases, the probability of error in transmission may be so much lower than the probability of error from other sources that error control may not be necessary. When very high accuracy is required, error-control techniques can be used effectively.

Error-detection and block-retransmission methods appear necessary in order to obtain a high degree of accuracy on long distance transmission. Forward-acting error-correcting codes may be used to improve the line efficiency when such methods are used.

It is possible to design around many of the data limiting characteristics of the network — the compandors and echo suppressors, for example. The variability in circuit characteristics can also be compensated for somewhat by corrective devices associated with either the data terminal equipment or, in some cases, the telephone channel itself. The compromise equalizers used for the 1200-bit-per-second tests are typical examples of what can be done in this direction.

For some applications, arrangements may be made to bypass certain facilities that limit the transmission of data signals. These may take the form of controlled access to the long distance switching network or perhaps the use of only certain telephone facilities and offices in the data service offering. In any case, the final decision as to the engineering design will be determined by the over-all economics.

The Bell System has a continuing effort to achieve higher speeds and greater accuracy, to provide more effective means for handling the variety of data transmission requirements and to broaden the scope of data processing applications by reducing the cost of transmitting information.

VIII. ACKNOWLEDGMENTS

The authors would like to express their appreciation and give special recognition to the following men in Bell Telephone Laboratories who have made major contributions to the success of the measurement program: L. F. Kelley and N. E. Snow, for the design and construction of the specialized test equipment; G. J. McAllister, for magnetic tape processing and computer analysis of error statistics; and L. F. Bugbee and H. J. Ewoldt, for computer analysis of transmission characteristics.

In addition, the enthusiastic cooperation of the many people in the operating companies of the Bell System, the American Telephone and Telegraph Company and Bell Telephone Laboratories is gratefully acknowledged.

REFERENCES

1. Duncan, J. A., Parker, R. D. and Pierce, R. E., Telegraphy in the Bell System, A.I.E.E. Trans., **63**, 1944, p. 1032.
2. Carter, C. W., Jr., Dickieson, A. C. and Mitchell, D., Application of Compandors to Telephone Circuits, A.I.E.E. Trans., **65** (Suppl.), 1946, p. 1079.
3. Clark, A. B. and Mathes, R. C., Echo Suppressors for Long Telephone Circuits, A.I.E.E. Trans., **44**, 1925, p. 481; Horton, A. W., The Occurrence and Effect of Lockout Occasioned by Echo Suppressors, B.S.T.J., **17**, 1938, p. 258.
4. Inglis, A. H. and Tuffnell, W. L., An Improved Telephone Set, B.S.T.J., **30**, 1951, p. 239.
5. Huntley, H. R., The Present and Future of Telephone Transmission, Elec. Engg., **75**, 1956, p. 686.
6. Wilkinson, R. I., Beginnings of Switching Theory in the United States, Elec. Engg., **75**, 1956, p. 796.
7. Pilliod, J. J., Fundamental Plans for Toll Telephone Plant, A.I.E.E. Trans., **71**, Part 1, 1952, p. 248.
8. Clark, A. B. and Osborne, H. S., Automatic Switching for Nationwide Telephone Service, A.I.E.E. Trans., **71**, Part 1, 1952, p. 245.
9. Truitt, C. J., Traffic Engineering Techniques for Determining Trunk Requirements in Alternate Routing Trunk Networks, B.S.T.J., **33**, 1954, p. 277.
10. Newell, N. A. and Weaver, A., Single-Frequency Signaling System for Supervision and Dialing Over Long Distance Telephone Trunks, A.I.E.E. Trans., **70**, Part 1, 1951, p. 489.
11. Newell, N. A. and Weaver, A., In-Band Single-Frequency Signaling, B.S.T.J., **33**, 1954, p. 1309.
12. Bogan, L. B. and Young, K. D., Simplified Transmission Engineering in Exchange Cable Plant Design, A.I.E.E. Trans., **73**, Part 1, 1954, p. 498.
13. Shaw, T., The Evolution of Inductive Loading for Bell System Telephone Facilities, B.S.T.J., **30**, 1951, p. 149.
14. Affel, H. A., Demarest, C. S. and Green, C. W., Carrier Systems on Long Distance Telephone Lines, B.S.T.J., **7**, 1928, p. 564.
15. Fisher, H. J., Almquist, M. L. and Mills, R. H., A New Single-Channel Carrier Telephone System, B.S.T.J., **17**, 1938, p. 162.
16. Starbird, L. C. and Mathis, J. D., Some Applications of the Type J Carrier System, B.S.T.J., **18**, 1939, p. 338.
17. Green, C. W. and Green, E. I., A Carrier Telephone System for Toll Cables, B.S.T.J., **17**, 1938, p. 80.
18. Elmendorf, C. H., Ehrbar, R. D., Klie, R. H. and Grossman, A. J., The L-3 Coaxial System, A.I.E.E. Trans., **72**, Part 1, 1953, p. 395.

19. Caruthers, R. S., The Type N-1 Carrier Telephone System: Objectives and Transmission Features, B.S.T.J., **30**, 1951, p. 1.
20. Edwards, P. G. and Montfort, L. R., The Type O Carrier System, B.S.T.J., **31**, 1952, p. 688.
21. Fracassi, R. D. and Kahl, H., Type ON Carrier Telephone, A.I.E.E. Trans., **72**, Part I, 1953, p. 713.
22. Roetken, A. A., Smith, K. D. and Friis, R. W., The TD-2 Microwave Radio Relay System, B.S.T.J., **30**, October 1951, p. 1041.
23. Weher, L. A., FM Digital Subset for Data Transmission Over Telephone Lines, Comm. & Electronics, no. 40, 1959, p. 867.
24. Edson, J. O., Flavin, M. A. and Perry, A. D., Synchronous Clocks for Data Transmission, Comm. & Electronics, no. 40, 1959, p. 832.
25. Nyquist, H., Certain Topics in Telegraph Transmission Theory, A.I.E.E. Trans., **47**, 1928, p. 617.
26. Lattimer, I. E., The Use of Telephone Circuits for Picture and Facsimile Service, Long Lines Dept., American Telephone and Telegraph Company, New York, 1948.
27. Mertz, P., Transmission Line Characteristics and Effects on Pulse Transmission, Proc. Symp. on Information Networks, Polytechnic Inst. of Brooklyn, April 1954.
28. Wood, F. B., Optimum Block Length for Data Transmission with Error Checking, Comm. & Electronics, no. 40, 1959, p. 855.
29. Brown, A. B. and Meyers, S. T., Evaluation of Some Error-Correcting Methods Applicable to Digital Data Transmission, I.R.E. Conv. Rec., Part 4, 1958, p. 37.
30. Hamming, R. W., Error Detecting and Error Correcting Codes, B.S.T.J., **29**, 1950, p. 147.
31. Hagelbarger, D. W., Recurrent Codes — Easily Mechanized, Burst-Correcting Binary Codes, B.S.T.J., **38**, July 1959, p. 969.